Micronet®
Faster and Easier Networks

User's Manual

IP Telephony Gateway, FXS + FXO Interface

Model No.: SP5012/S, SP5014/S

Website: http://www.micronet.info

Steps in configuration

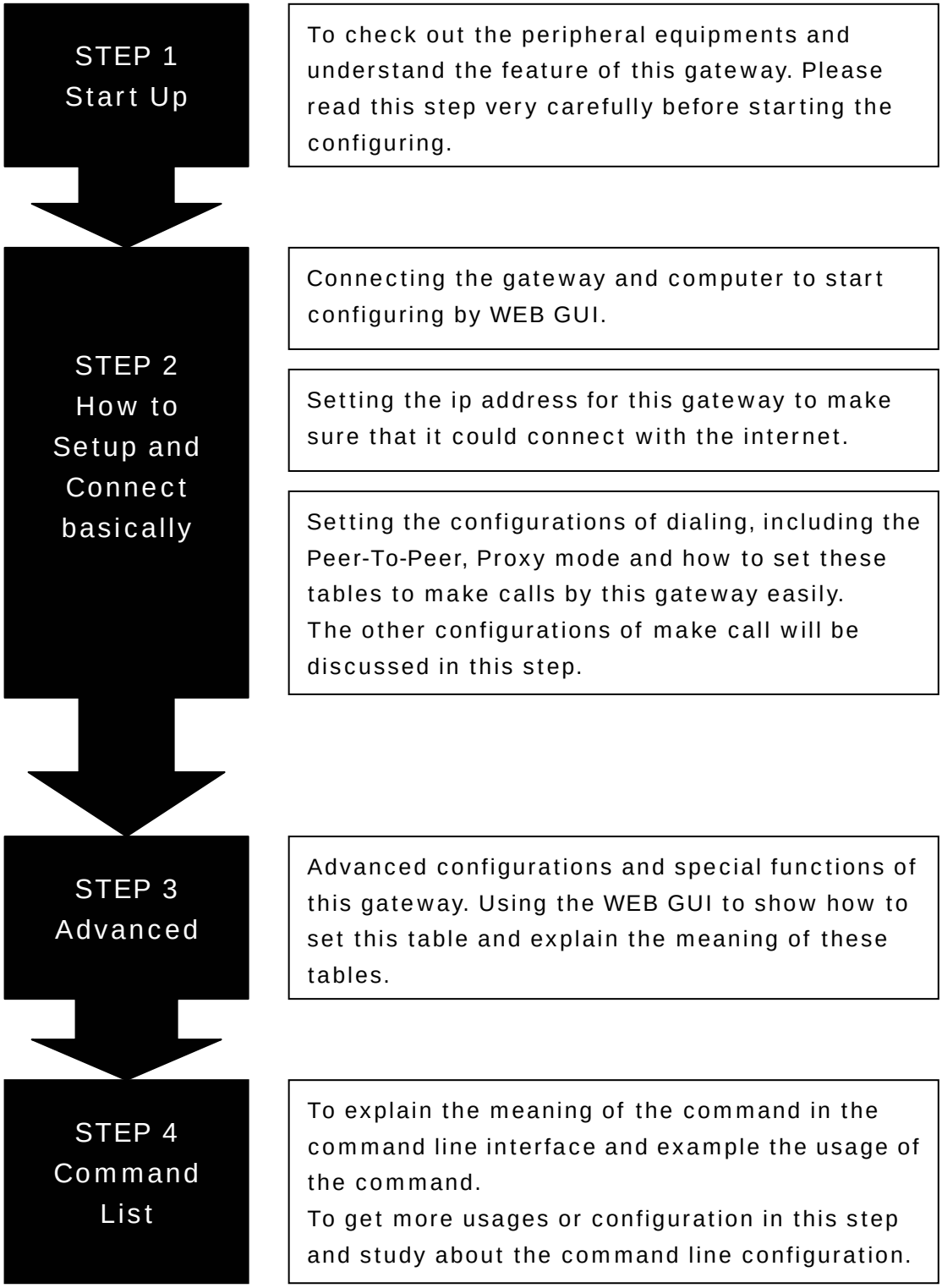


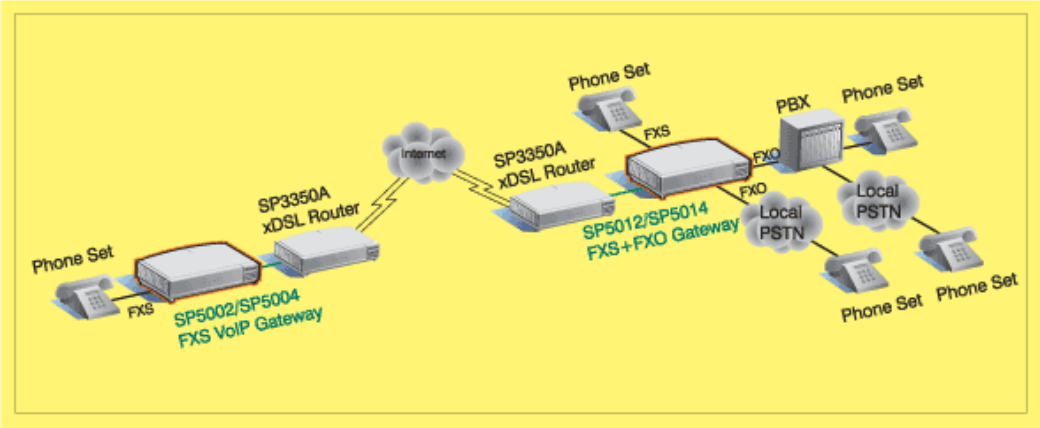
Table of Contents

1 Start Up

1.1 Introduction

Micronet SP5012/S,SP5014/S is a multi-port FXS+FXO gateway. It supports an innovative intelligent call routing function that transparently routes calls to destination either through PSTN or Internet.
Micronet SP5012/S,SP5014/S provides voice over IP and FAX over IP services for Internet Telephony Services Provider (ITSP/ISP) and Office/SOHO IP-PBX application.

Application Architecture



- **FXO ports can connect with PSTN Line or Extension Line of PBX**
- **FXS ports can connect with Phone Set or Trunk Line of PBX**

1.2 Features and specification

Features

- IETF RFC 3261
- Automatically Dial Path Selection (IP or PSTN)
- PSTN Line switch to telephone set when power is failure
- PPPoE support
- Behind NAT router or IP sharing device
- DNS server inquiry
- Provide Peer-to-Peer Mode (Non SIP Proxy needed) selection
- E.164 Dial Plan
- TFTP/FTP software upgrade
- Remote configuration/ reset
- LED indication for system status
- Support Fix IP, DHCP and PPPoE

Audio feature

- Codec -- G.711 a/μlaw, G.723.1 (6.3kbps), G.729, G.729A
- G.168/165-compliant adaptive echo cancellation
- Dynamic Jitter Buffer
- Completed voice band signaling support
- Provide In-band or RFC2833 DTMF generation/detection
- Provide call progress tone

Management Feature

- TELNET/Console port and Web Browser configuration

Certification

- UL, CE, FCC

FXS Features

- 2-wire loop start
- Programmable On-Hook voltage, Ring voltage/Cadence/Frequency, Loop current
- Line polarity reversal generation

FXO Features

- 2-wire loop start
- Support auto-attendant (Tone or voice greeting)
- PSTN polarity reversal detection
- Provide 2nd dial tone to PSTN
- Disconnect tone detection
- Asking ping function with the incoming calls from PSTN side
- Record and analyze the Tone from PSTN side

Environmental

- Operation temp:0°C to 40°C
- Humidity: 10% to 90% (Non-condensing)

1.3 Accessories and equipment

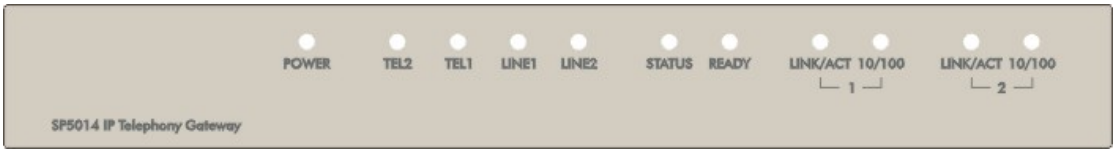
- The voice gateway in 2 FXS and 2 FXO ports or 1 FXS port and 1 FXO port models and two RJ-45 connector (WAN and LAN).
- The AC adapter.
- The CD of User’s Manual.
- The connection cable in RS-232 interface.

1.4 Appearance

Front panel: The LED lights provide related system messages of the gateway.

SP5012/S

SP5014/S



Power: Light on means Gateway is power on, and vice versa.

TEL: Light on means the line is in use (off-hook), and vice versa.

LINE: Light on means the line is in use (off-hook), and vice versa.

Status:

1. LED light on means Gateway has successfully registered to Proxy when it is in the Proxy Mode.
2. LED flash means Gateway is not registered to the Proxy when it is in the Proxy Mode.
3. Or when Gateway is in downloading mode, LED should be flash as well.
4. LED light off means Gateway is in Peer-to-Peer Mode.

Ready:

1. Light on and slow flash means Gateway is in normal mode.
2. Light on and fast flash means Gateway is in downloading mode.

WAN: Connected to Public Ethernet

1. Line- LED light on means Gateway is physically connected to the Ethernet correctly.
2. ACT- LED light on and flash when Ethernet data is being transmitted / received.

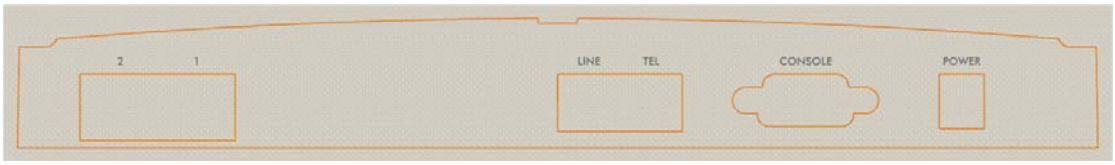
LAN: Switch to another device, such as PC

1. Line- LED light on means Gateway is physically connected to the Ethernet correctly.
2. ACT- LED light on and flash when Ethernet data is being

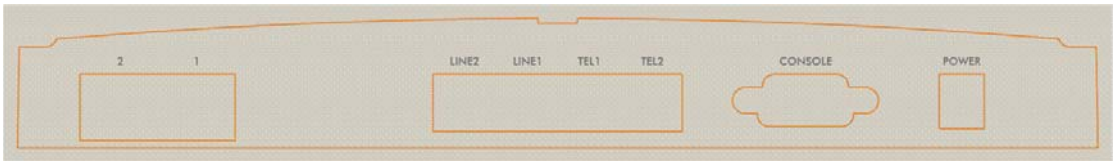
transmitted received.

Back panel:

SP5012/S



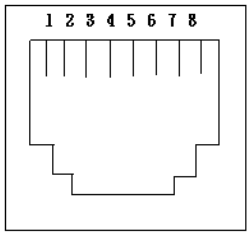
SP5014/S



1. **Ethernet Port**

   LAN/WAN: 10/100 Base-T; RJ-45 socket, complied with ETHERNET 10/100base-T.

   The pin-out is as following:



PIN 1, 2: Transmit

PIN 3, 6: Receive

2. **COM:**

   RS232 console port (DB-9pin **male** connector)

   Note: use straightforward cable to connect to your computer.

PINOUTS

Pin	Name	Dir	Description
2	RXD	←	Receive Data
3	TXD	→	Transmit Data
5	GND	—	System Ground

3. **TEL:**
   RJ-11 connector, FXS interface is for connecting the analog phone sets or trunk line of PABX.
4. **LINE:**
   RJ-11 connector, FXO interface is for connecting the extension line of PABX or PSTN Line.
5. **12V DC:**
   Input AC 100V~120V; output DC12V.

2 How to Setup and Connect basically

2.1 System Requirement

1. One PC (a) Pentium 100 or above, 64 RAM, Windows 98 or above.
   (b) Ethernet card or COM port
2. One standard straightforward RS-232 cable (female connector to Gateway side).
3. Analog telephone sets or the PBX trunk Lines.
4. PBX extension Lines or PSTN Lines.
5. Software tools – Hyper Terminal, TELNET, Web Browser.

2.2 IP Environment Setting

User must prepare a valid IP address, complied with IP Network, for Gateway's proper operation.

For testing the validation of chosen IP address, using the same IP configuration in other PC or Notebook, and then try to connect to Public Internet (go to well-known website, receive Internet mail, or ping a specific public IP address). If it works, use the same IP address and network configuration for Gateway.
Please follow up the step for the configuration of your computer or notebook.

For Windows 2000/NT

Please make sure that the network interface of your computer is working fine and the cross over line (RJ-45) is connecting with the computer correctly or you could use a hub to connect with your computer and this gateway. Turn on your computer and configure the network parameter as follow:

1 Go to the **start** menu and enter the **setting** area. Click **control panel**.

2 Enter the network configuration.

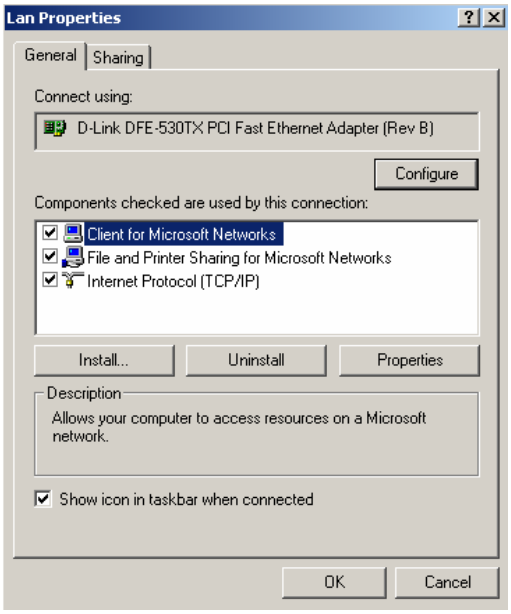


Figure 2.1: Network Configuration

3 Select the **Property** of the LAN card.

4 Setup the ip address, subnet mask and default gateway as below:

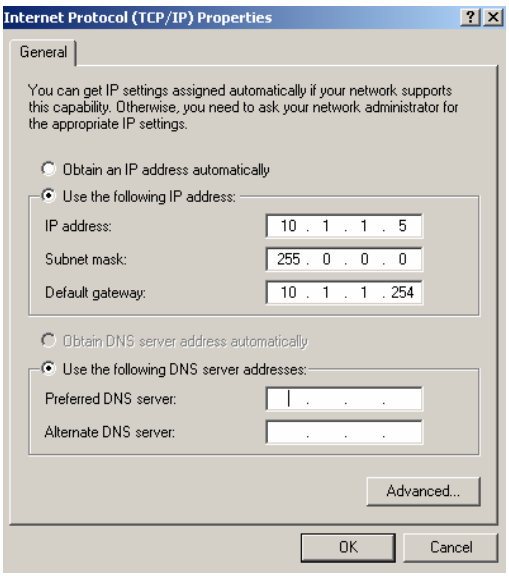


Figure 2.2: Configure the network

5 Click OK after you finished the network setup.

The default ip address, netmask and default gateway address of the gateway is 10.1.1.3, 255.0.0.0, 10.1.1.254.

2.3 Network configurations in your gateway

1 Key in the ip address of the gateway (http://10.1.1.3) with the browser. (see figure 2.3)

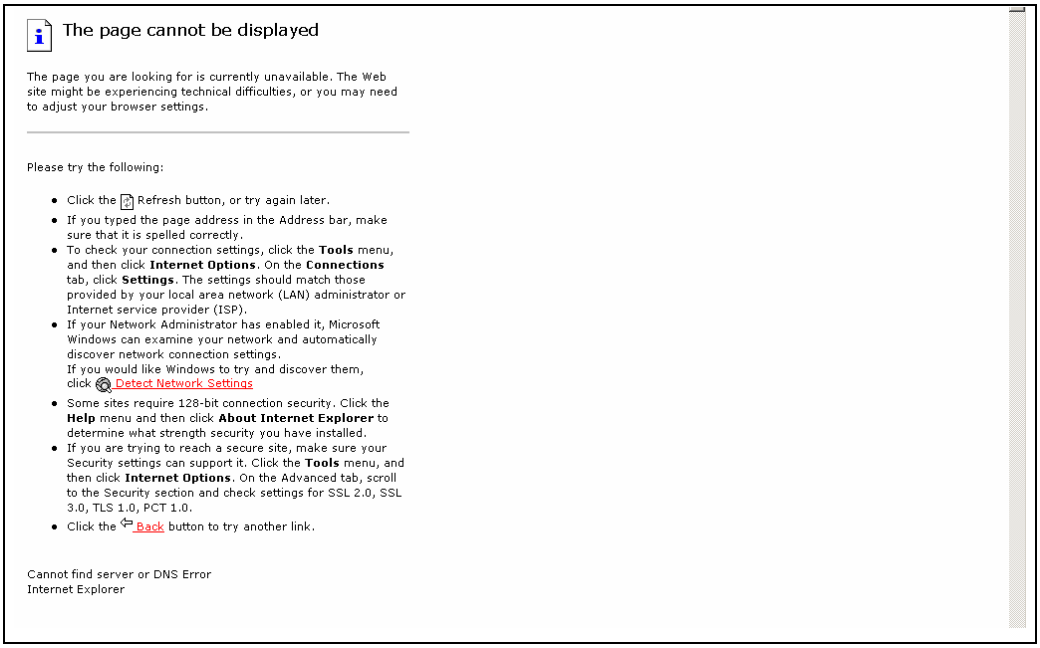


Figure 2.3: WEB Browser

2 After key in the ip address, you have to enter the user name and password to enter the WEB configuration. (Username: root ; No password) (see figure 2.4)

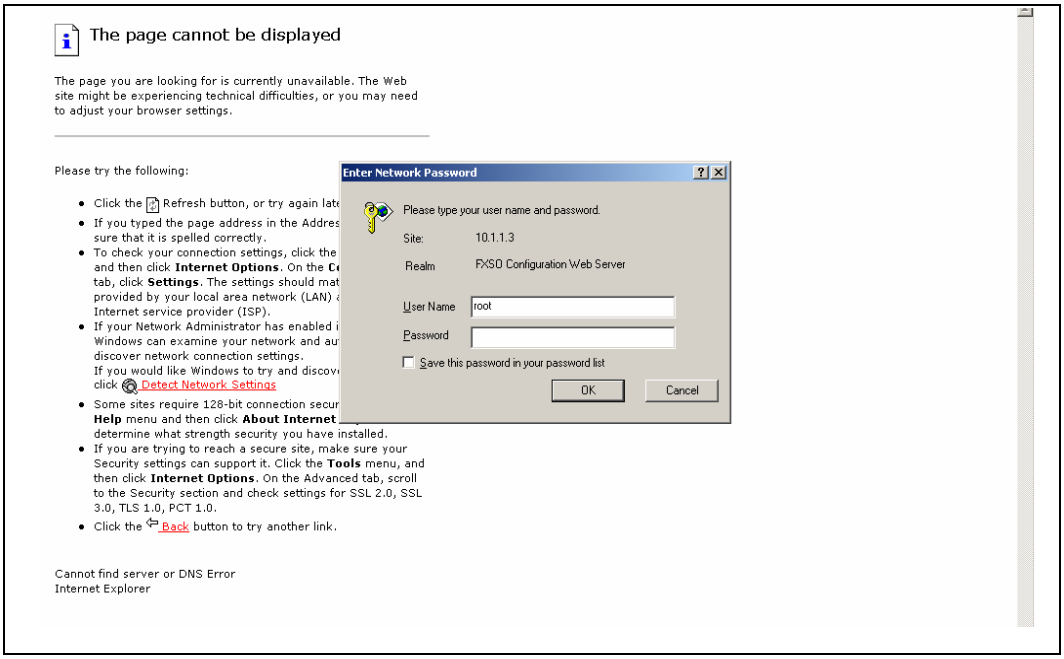


Figure 2.4: Login the username and password

3 You will enter the main page of the configuration after key in the login name and password correctly: (see figure 2.5)

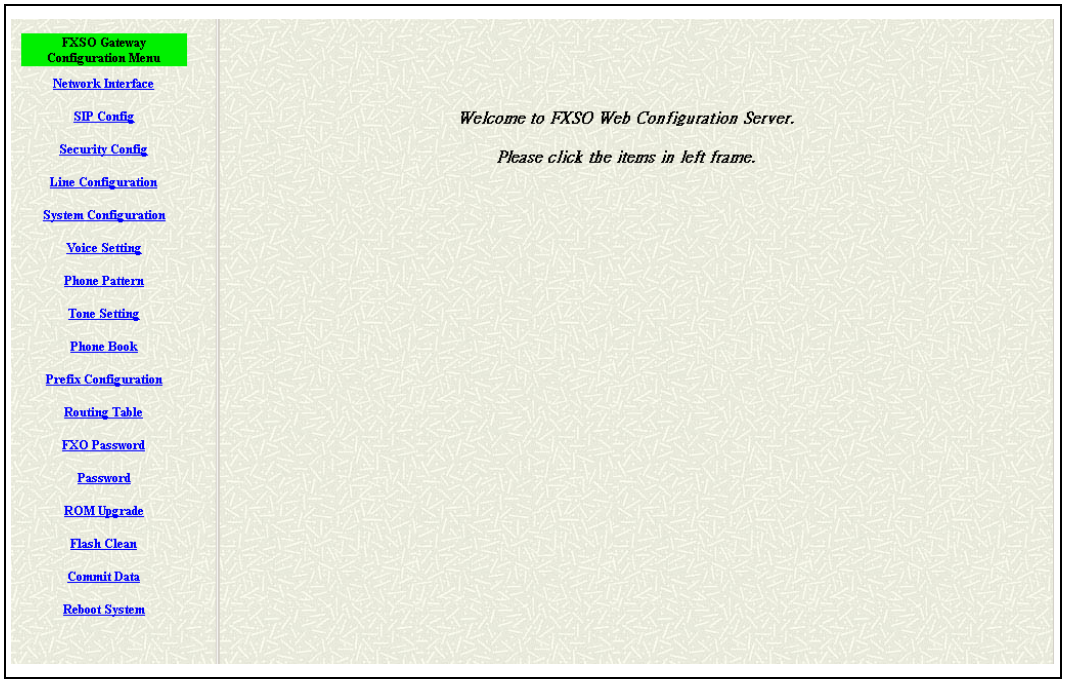


Figure 2.5: The main WEB configuration

4 Press the **Network Interface** to configure the networking of your gateway. (see figure 2.6)

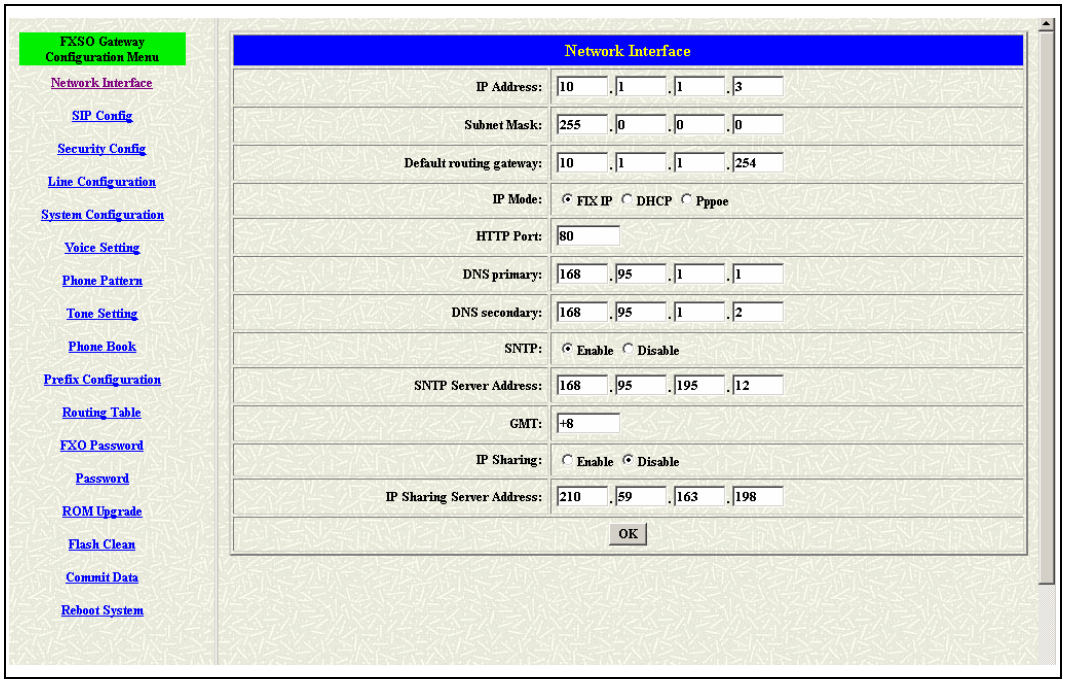


Figure 2.6: The Network Interface

2.3.1 Static IP address

1 Please get the correct IP address, netmask and default gateway address from your ISP first. Press the OK button if you finished. (see figure 2.7)

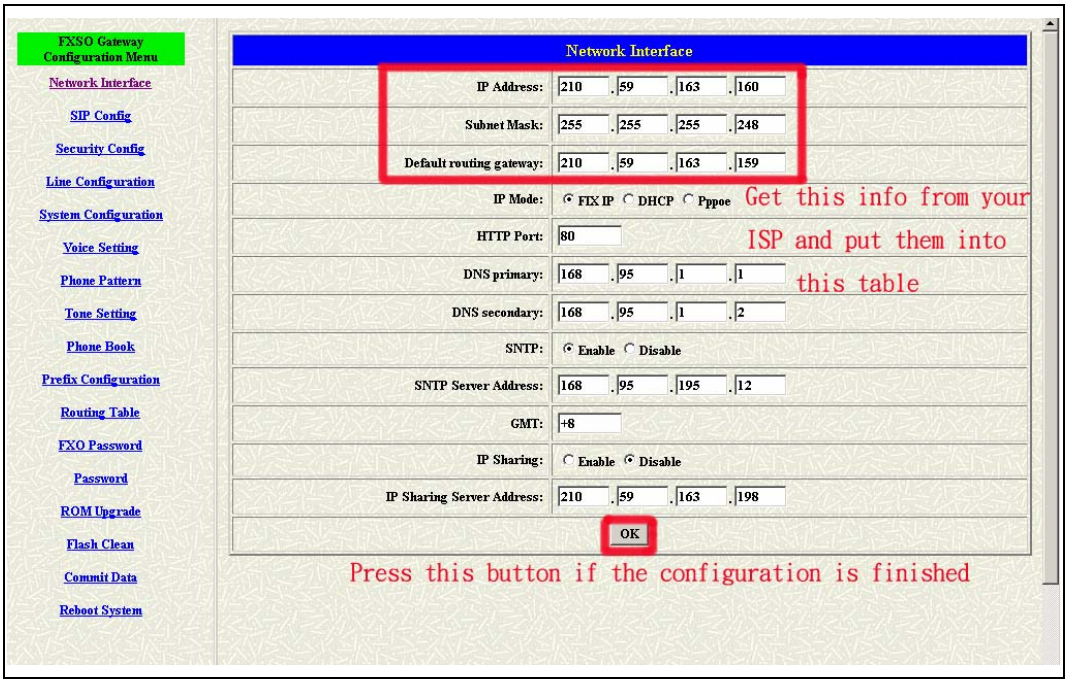


Figure 2.7: Configure the static ip address

2 Press the commit if you finish the configuration. (see figure 2.8)

Figure 2.8: Commit the data

3 Press the reboot if you want the configuration executed. (see figure 2.9)

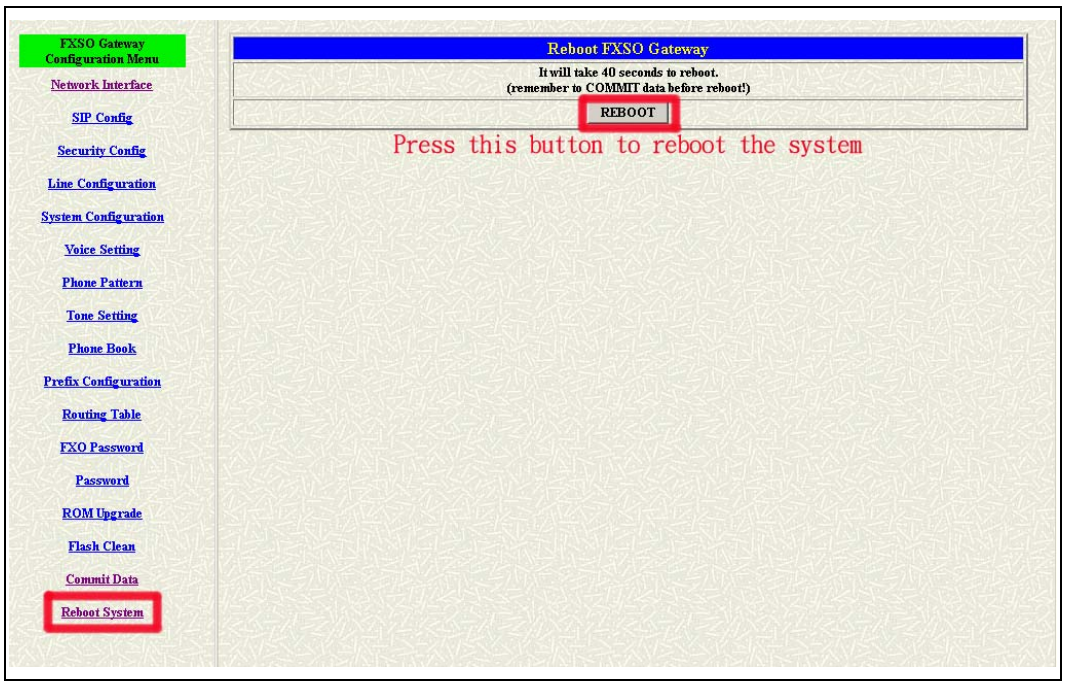


Figure 2.9: Reboot the system

2.3.2 DHCP mode

1 Enable the DHCP if you are using the cable modem or DHCP server. (see figure 2.10)

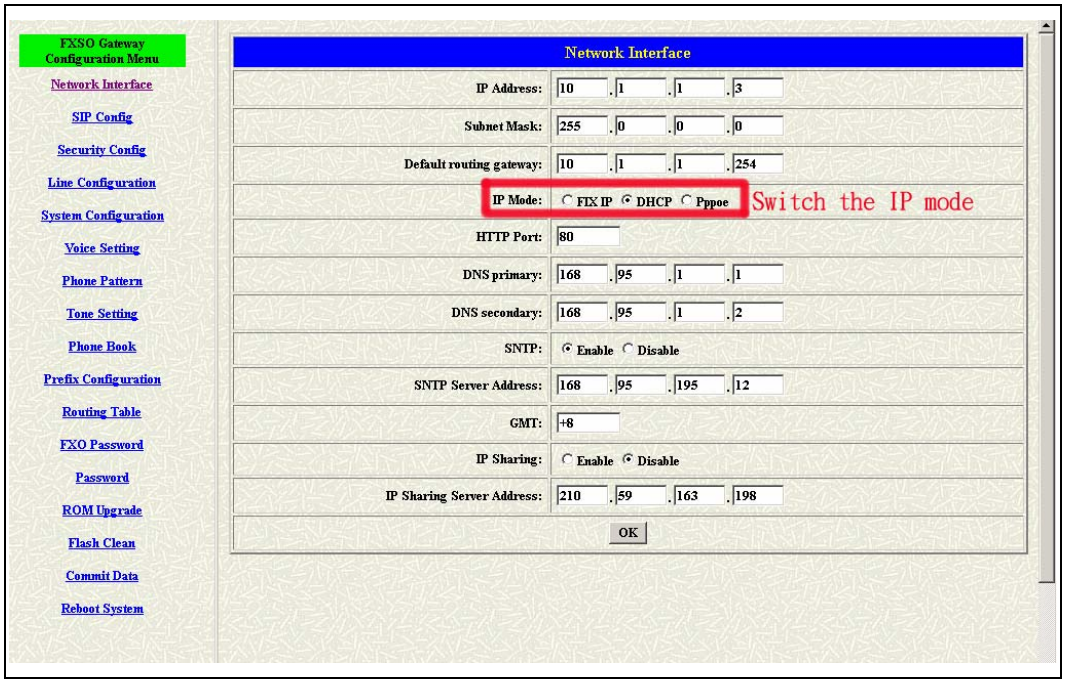


Figure 2.10: Enable the DHCP function

2 Please commit the data and reboot the machine after you enable the DHCP function.

2.3.3 PPPoE mode

1 Switch to the PPPoE mode and press the “OK” button. Press the **Network Interface** button after the “OK” button. (see figure 2.11)

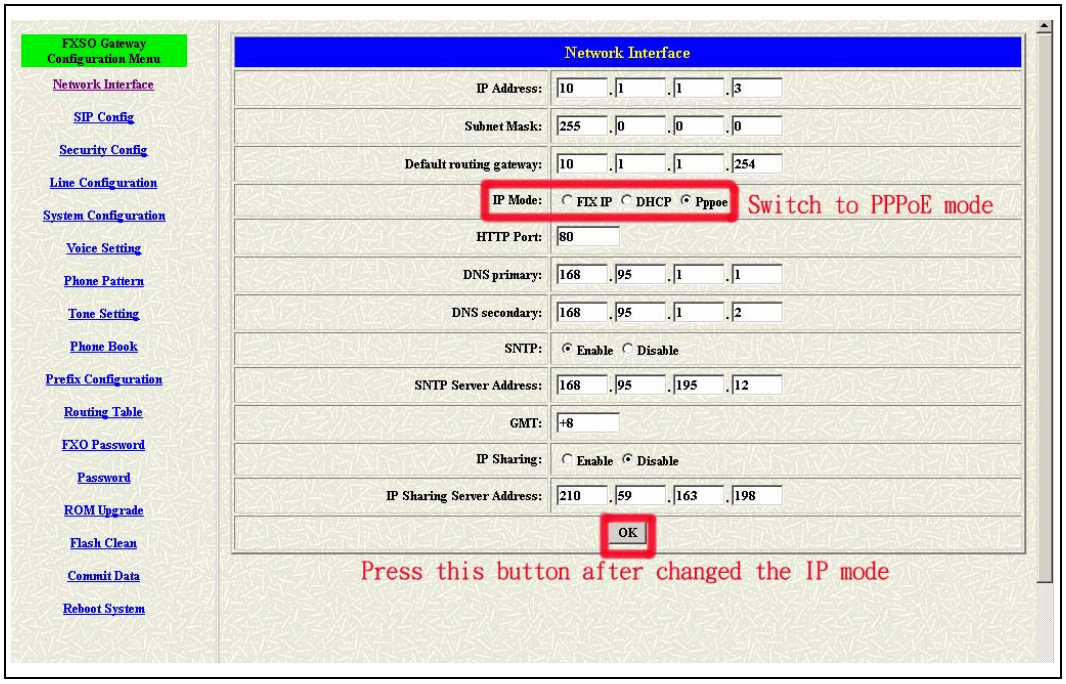


Figure 2.11: Switch to the PPPoE mode

2 Enter the Login account and password. Press the “OK” button if the configuration is finished. (see figure 2.12)

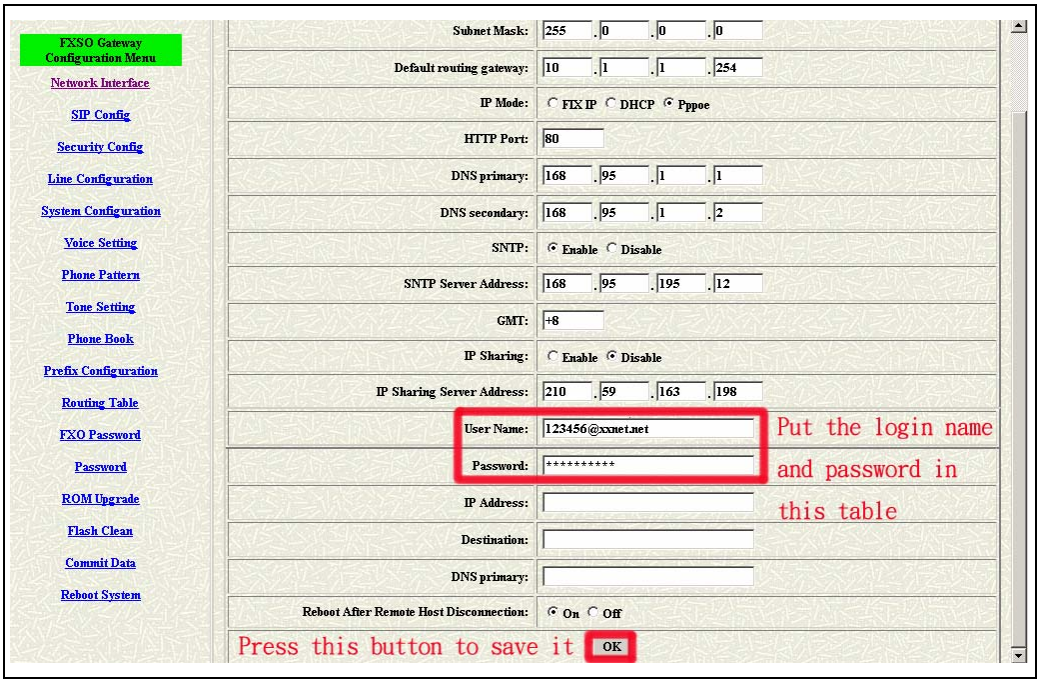


Figure 2.12: Enter the Account and password

2 Please commit the data and reboot the machine after you finished the configuration about the PPPoE function.

2.4 Making a VoIP Call

There are two modes that you could configure the gateway for making VoIP calls. One is the Peer-to-Peer mode, another is Proxy mode. The configurations and functions are different. Please make sure about the mode you want and follow up the step to configure your gateway.

2.4.1 Configure the gateway into the Peer-to-Peer mode

1 Enter the SIP Configuration table and change the mode to Peer-to-Peer. Define the port numbers whatever you like. Press the “OK” button if the configuration is all finished. (see figure 2.13)

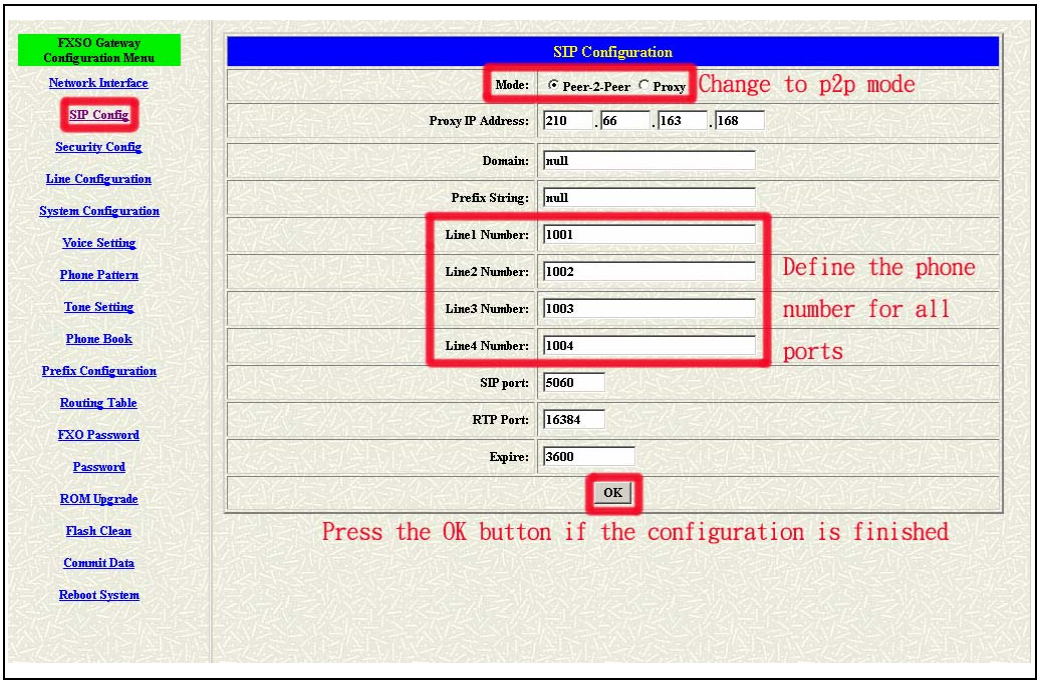


Figure 2.13: Configure the Peer-to-Peer mode

2 Enter the Phone Book configuration table and configure the name, ip address and phone number of the destination. (see figure 2.14)

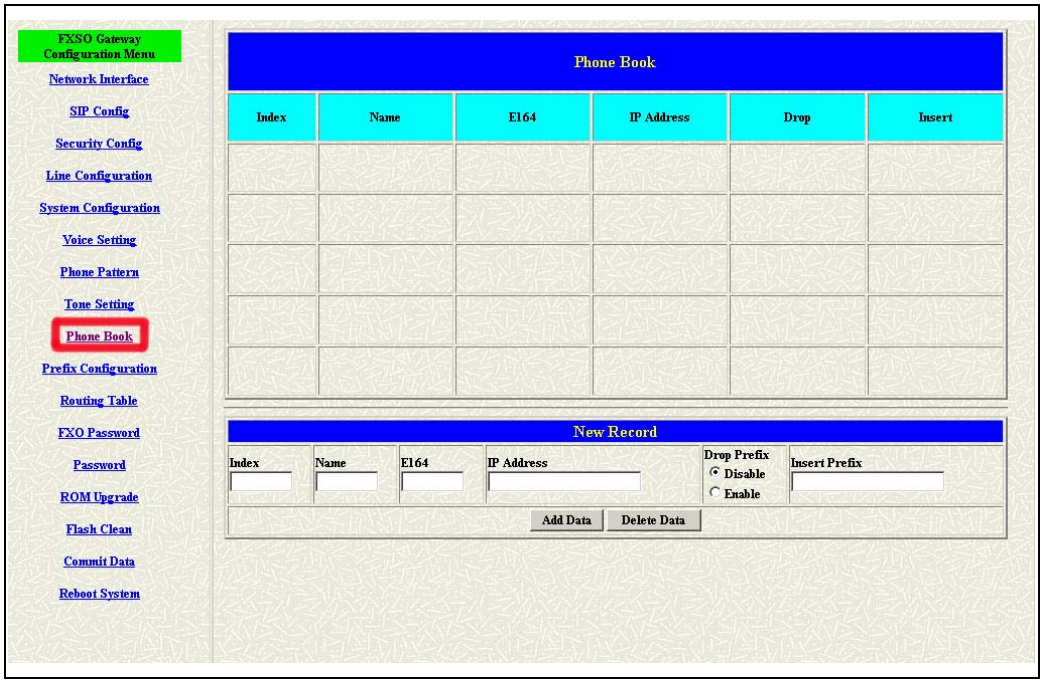


Figure 2.14: Phone Book

【Example】

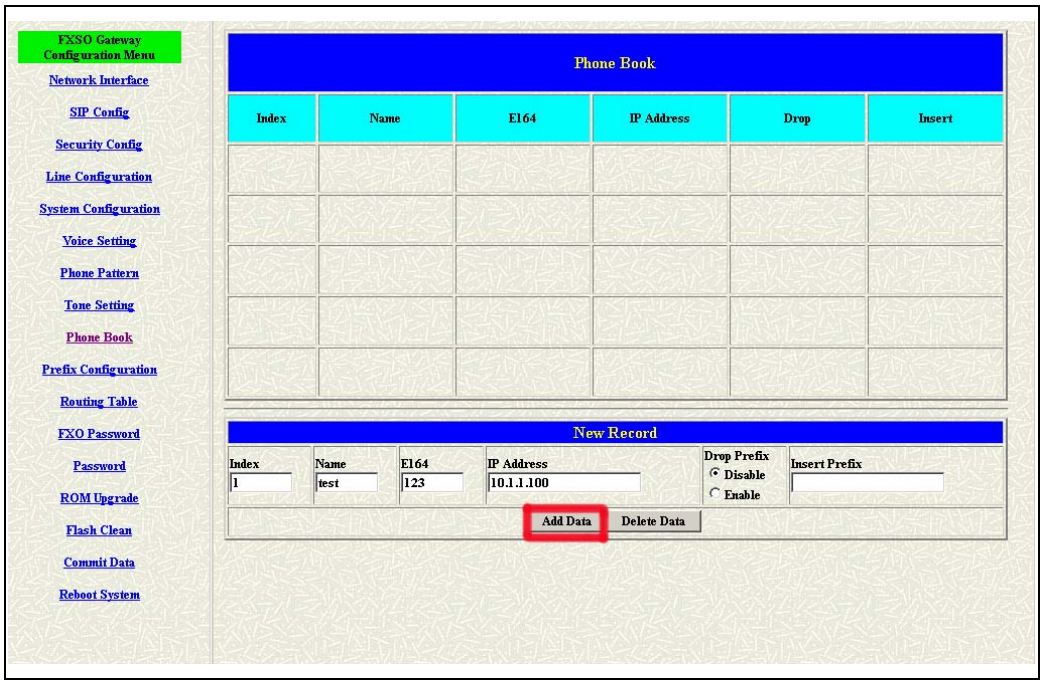


Figure 2.15: The example of Phone Book configuration

The name of the destination: **test**

The E164 number (phone number) of the destination: **123**

The ip address of the destination: **10.1.1.100**

The call signal port of the destination: **1720**
(The port will be 1720 if you don' t define it)
Drop prefix: **Enable – The e164 number you define will be deleted**
Disable – The e164 number you define will be kept
Insert prefix: **To add a number you define in this table**

Press the "Add Data" button when you finished, and the new table will display on the first index if you press the Phone Book configuration button.

4 Please Commit it and Reboot the system if the configuration is finished.
(see figure 2.16)

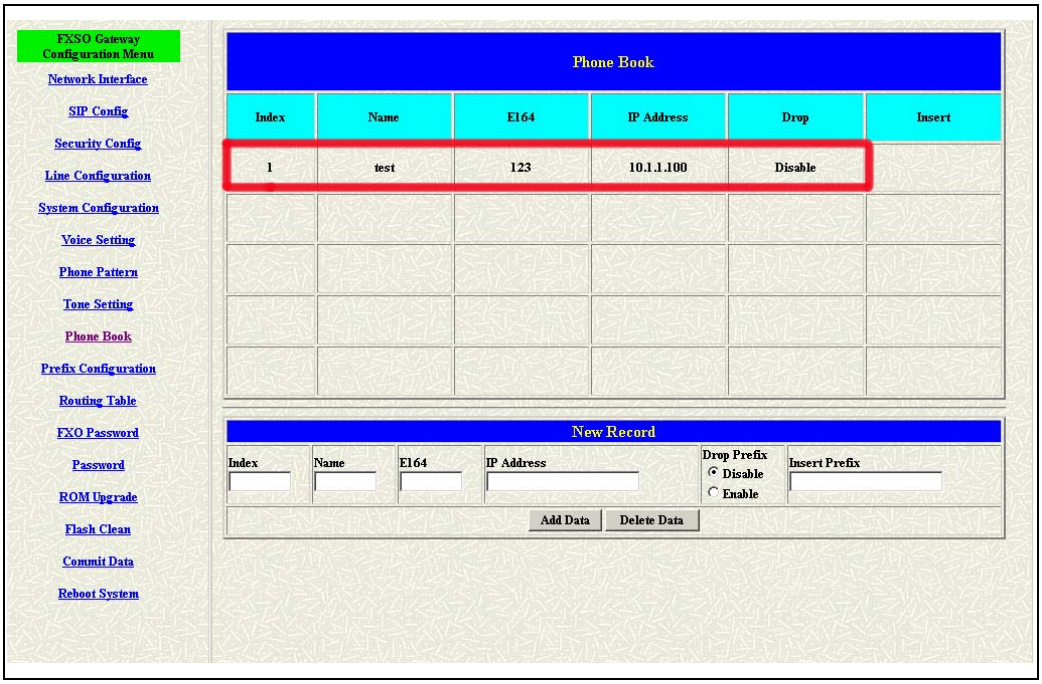


Figure 2.16: To show the Phone Book record

- **Phone Book is only for the Peer-to-Peer mode.**
- **Fifty index supported.**

【The application in the drop and insert function】

Input (E164)	Drop	Insert	Output
100	Disable	X	100
200	Disable	0	0200
300	Enable	X	X
400	Enable	500	500

※ X – Do not enter any numbers

2.4.2 Configure the gateway into the Proxy mode

1 Enter the SIP Config table and change the mode from Peer-to-Peer to Proxy.

To change the Proxy information from your service provider (Ex: The Proxy IP, Domain and Line numbers). (see figure 2.17)

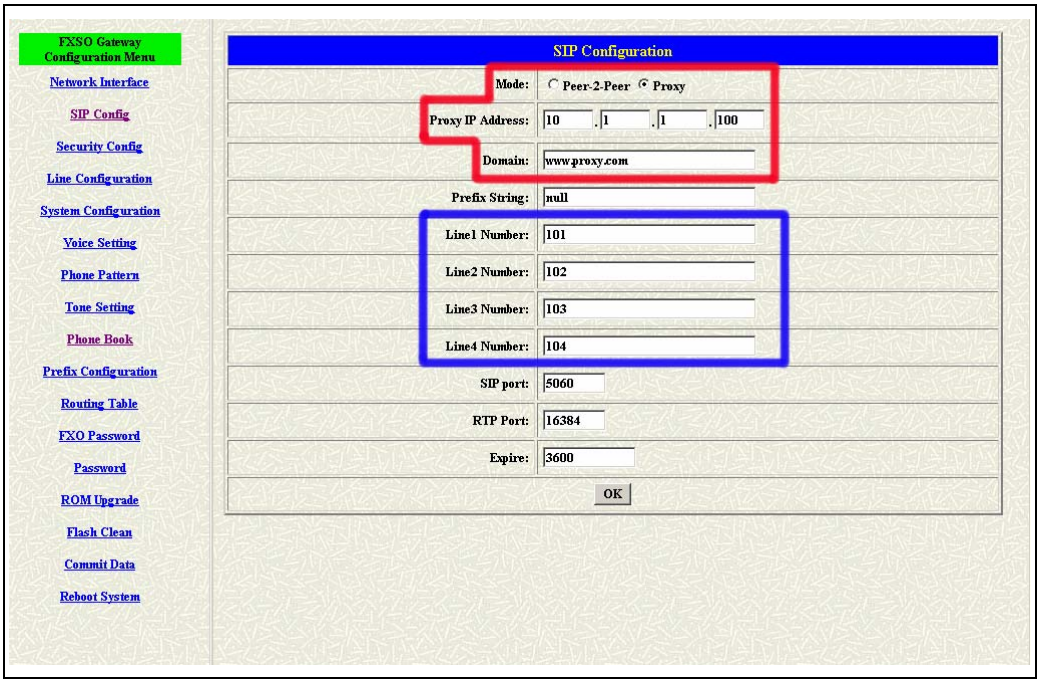


Figure 2.17: Configure the Proxy info

2 Press the OK button that is on the bottom of this page to save the

configuration.

3 Switch to the Security Config page and put the user account and password in the correct table. Please get this info from your ITSP. Press the OK button if the configuration is finished. (see figure 2.20)

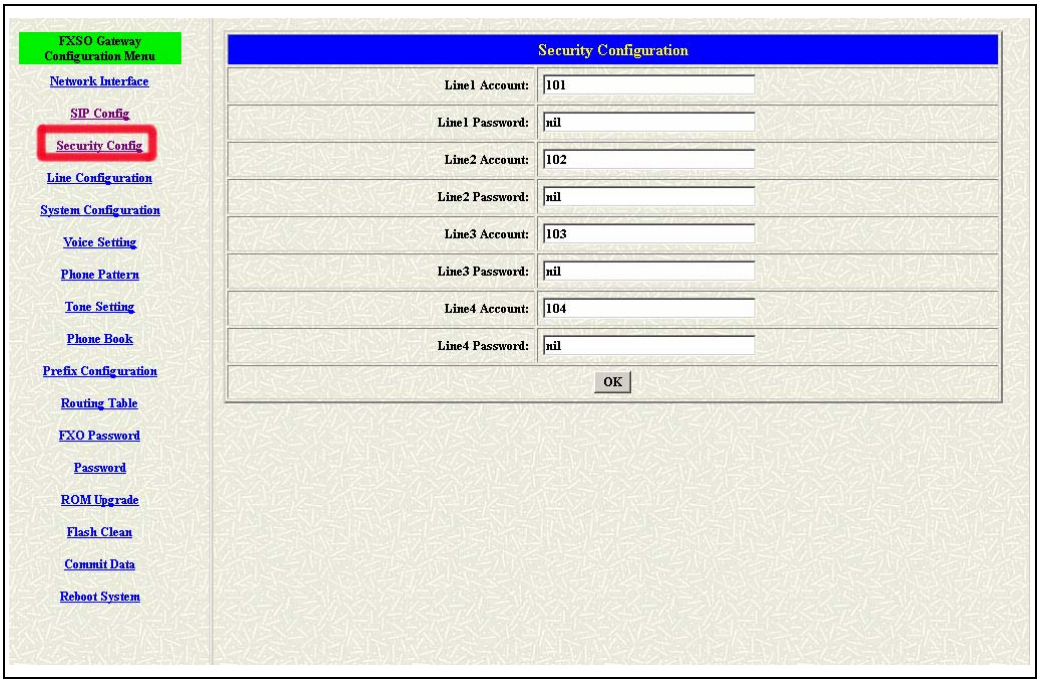


Figure 2.20: Configure the Security info

4 Press the Commit Data and Reboot System buttons when you finished the configuration.

3 Advanced

There are too many advanced commands for the advanced users. The following chapters are based on the application layer. Please get the info what you need. If you need the command, please watching the chapter of Command Line Interface.

3.1 Line

The Line configuration will show the status of the registrations and the ports. It includes the hunt group, hotline, and no answer forward configuration. Press the Line configuration button to enter configuration table (see figure 3.1)

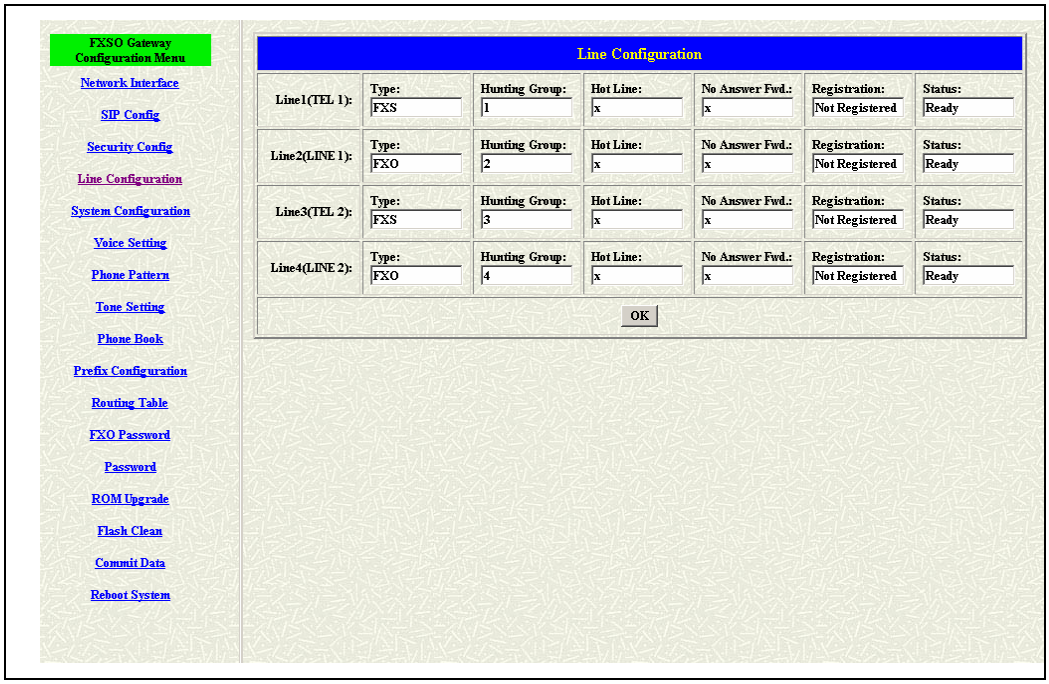


Figure 3.1: Line Configuration

- Type – Show the type of this port. There are only two types of this gateway. One is FXS type another is FXO type. It couldn't be changed.
- Hunting Group – Define the group number of this port. When the port is busy, the call could be transferred to another port in the same group.
- Hotline – Enable or Disable the hotline mode. The hotline mode will be enabled if you enter the hotline number. The default setting is disabled.
- No Answer Forward – When the port didn't answer the call, this call will

be forwarded to the number you configured. This is only for the E164 number or the phone numbers you want to transfer.

This function should be used with the forwardtime command (in the System configuration table). The range of the forward time is 5 to 65535. If the function is enabling, the No Answer forward number is 123, and the forward time is 5 seconds. The call will be forwarded to the destination with 123 numbers when the origin port didn't answer the call.

- Registration – To show the gateway registered on the Proxy Server or not.
- Status – To show the port is busy or ready.

3.2 System Configuration

There are some parameters in the system configurations, please get more detail as following. (see figure 3.2)

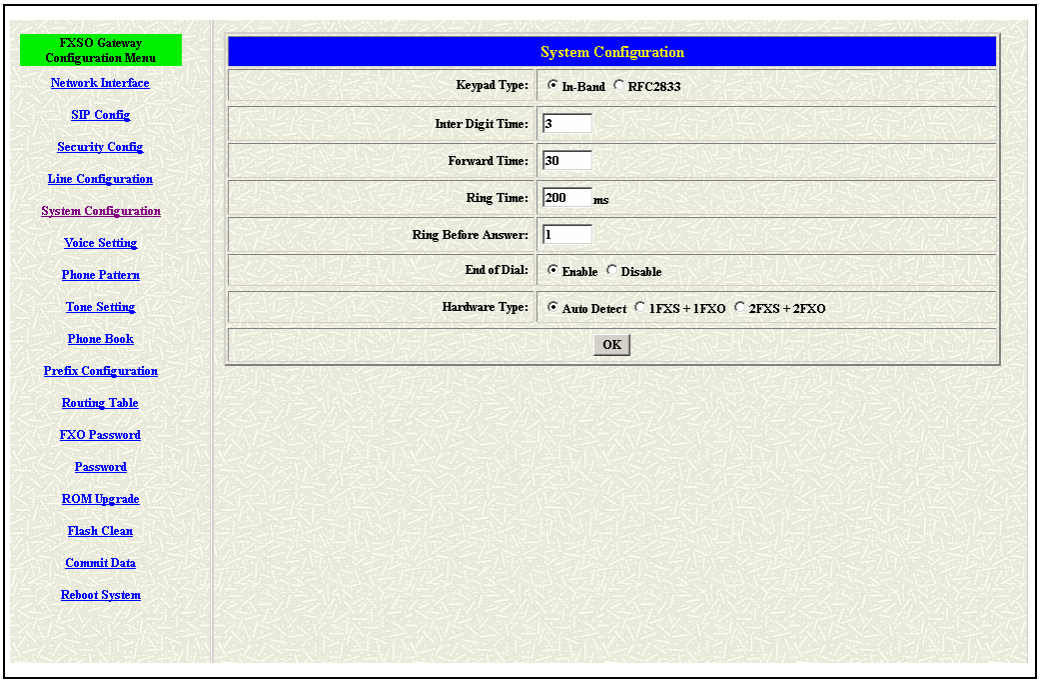


Figure 3.2: System Configuration

- Keypad type – There are tow types for the Keypad. On is the In-Band type, another is the RFC2833 type (Out-Band). User could define the keypad type for the dialing.
- Inter Digit Time – It's the time for the time out during the dialing numbers.
- Forward Time – It's the time for the no-answer-forward. Users have to configure it with the no-answer-forward function.
- Ring Time – FXO will detect the ring tone according this time.
- End of Dial – It will transfer the digit "#" if this function is disabling.
- Hardware Type – It's for the hardware issue.

3.3 Voice Setting

User could define some parameters about the voice in this voice-setting page. (see figure 3.3)

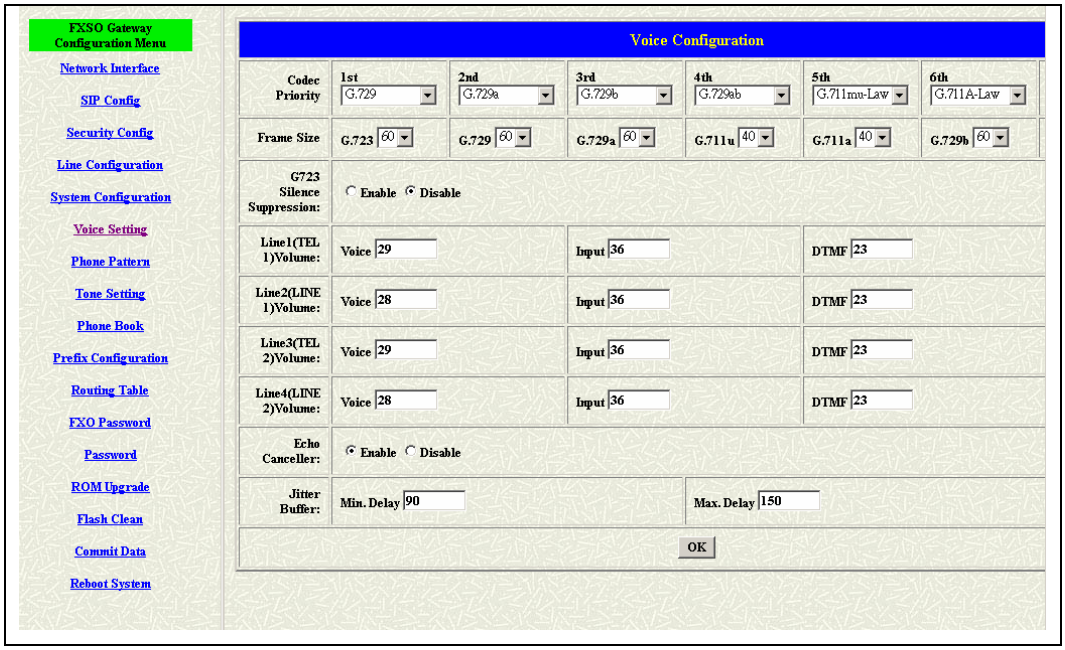


Figure 3.3: Voice Setting

- Codec Priority：It's for the codec setting. User could use the codec, which they want by the setting.

There are two firmware versions for SP5014/S. One is for G.723 only, another is for G.729 only. It with no meanings if users use the G.723 version and configure the codec in G.729. User could check out the firmware version first and configure the codec second. SP5012/S could support all the codec.

- Frame Size：It's the packet size for all codec. It will take more bandwidth if users configure the packet size in the minimum value.
- G723 Silence Suppression：For the VAD and CNG function support.
- Volume：To adjust the gain of the output, input and DTMF.
- Echo Canceller：To enable the echo cancellation function.
- Jitter Buffer：To adjust the Jitter Buffer size to avoid the packets losing.

A large jitter buffer causes increase in the delay and decreases the packet loss. A small jitter buffer decreases the delay but increases the packet loss. The size of the jitter buffer depends on the condition of the network, which varies with time. Typically the packet loss should be less than 10% for a good quality of speech.

3.4 Phone Pattern

The FXSO could generate some tones, such like the busy tone, dial tone, ring back tone and second dial tone…etc. Users could adjust these tones or get the detail info from this page. (see figure 3.4)

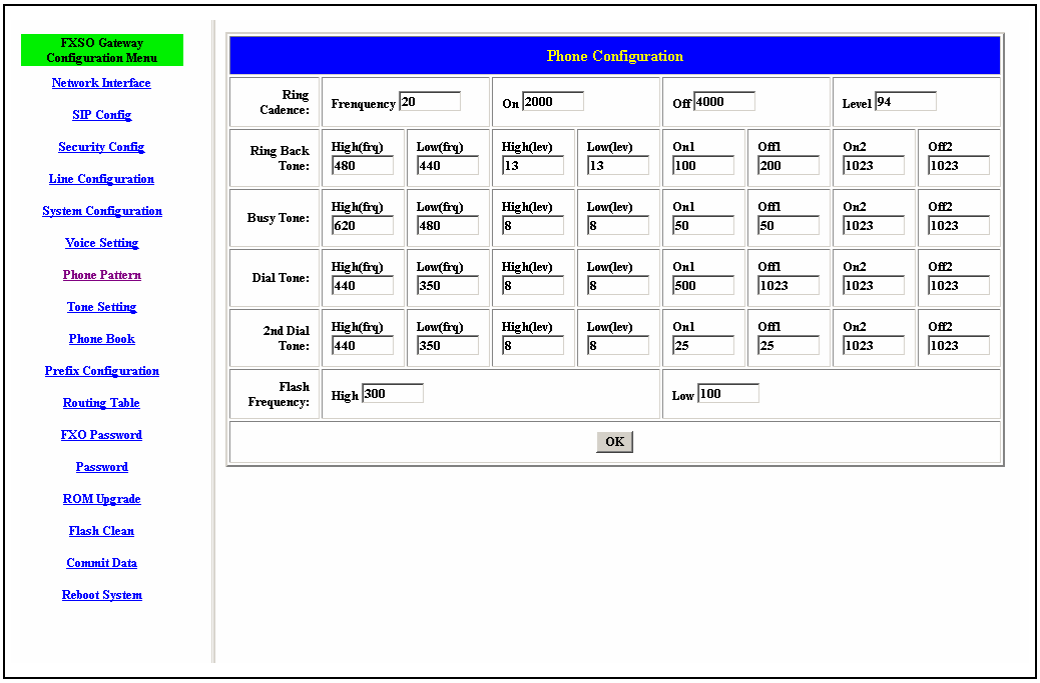


Figure 3.4: Phone Pattern

- Ring Cadence – Adjust the pattern for the cadence of the Ring tone. Including the Frequency, On time, Off time and the gain level.
- Ring Back Tone – Adjust the pattern for the Ring Back Tone. Including the High, Low frequency, High, Low Level, and the On, Off time.
- Busy Tone – Adjust the Busy Tone.
- 2nd Dial Tone – Adjust the second dial tone.
- Flash Frequency – Adjust the High, Low frequency for the Flash.

3.5 Tone Setting

The Tone Setting is for the Tone detecting. The call will be dropped if the pattern of the tone from PSTN side is as same as the pattern in the disconnect tone table. The same result for the Ring Back Tone. User could define the pattern of the disconnect tone if the disconnect tone from PSTN side is not the standard tone. (see figure 3.5)

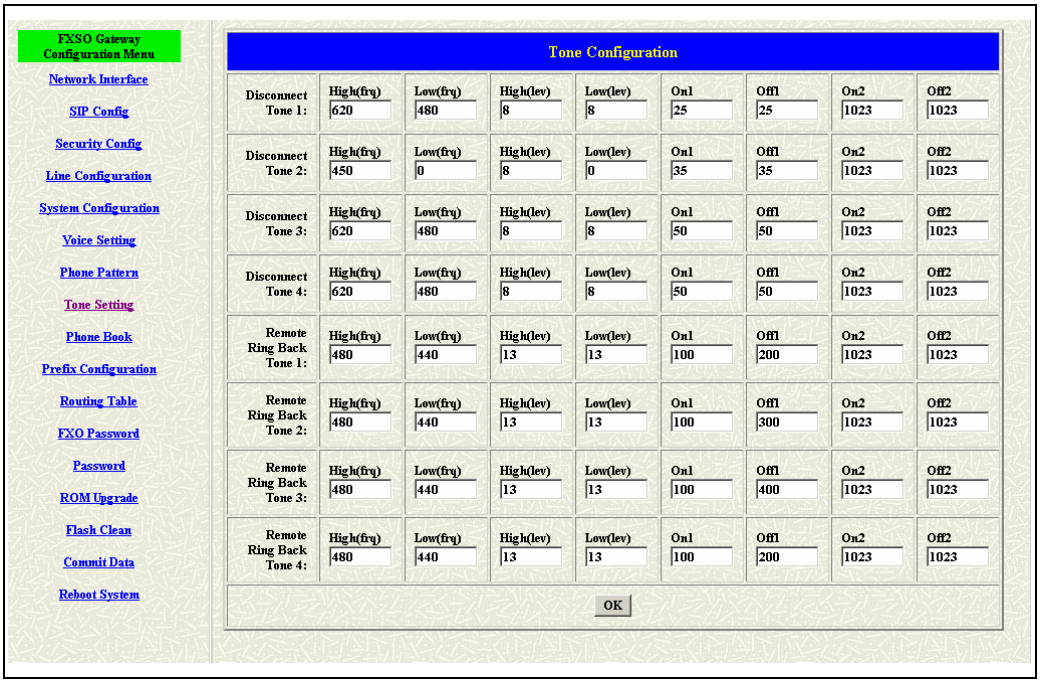


Figure 3.5: Tone Setting

- Disconnect Tone – Users could put the correct pattern of the disconnect tone in this table. The call will be dropped if the tone from PSTN side is match with these patterns. Users could have four tables for the disconnect tone.
- Remote Ring Back Tone – User could adjust this table to help the FXSO gateway to detect the Remote Ring Back Tone. There could be four tables for the configuration.

3.6 Prefix

The Prefix function is using the drop and insert function (see figure 3.6).

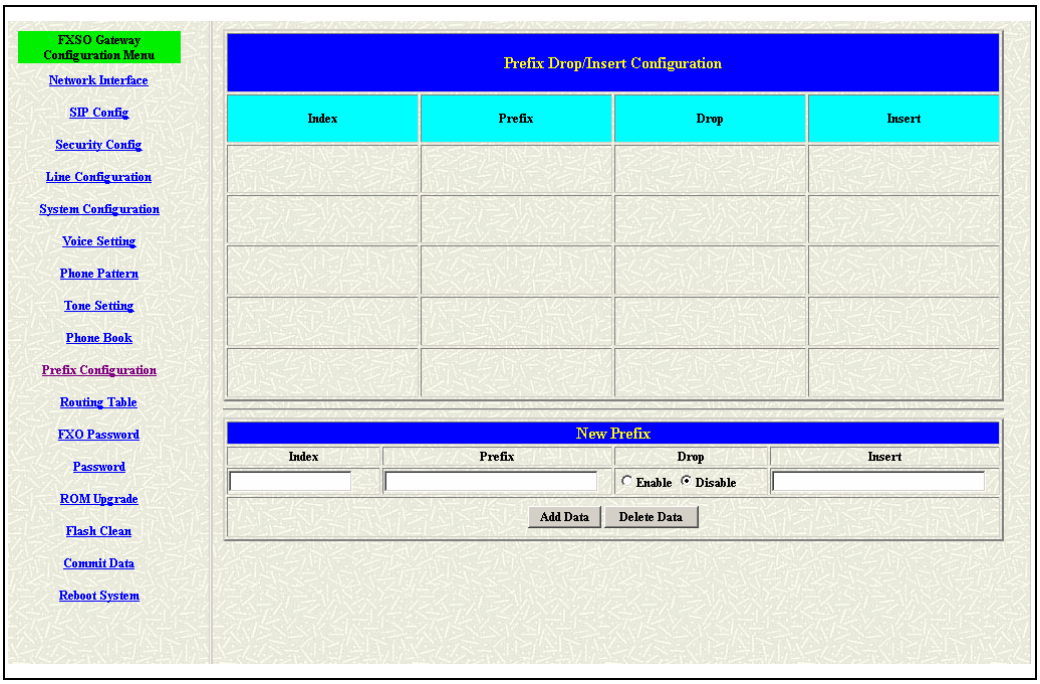


Figure 3.6: Prefix Configuration

There is a rule between Prefix and Routing command, the Prefix command have the higher priority over the Routing command. If there is an incoming call from any sides, the Routing will check this calling number after the Prefix checked (see figure 3.)

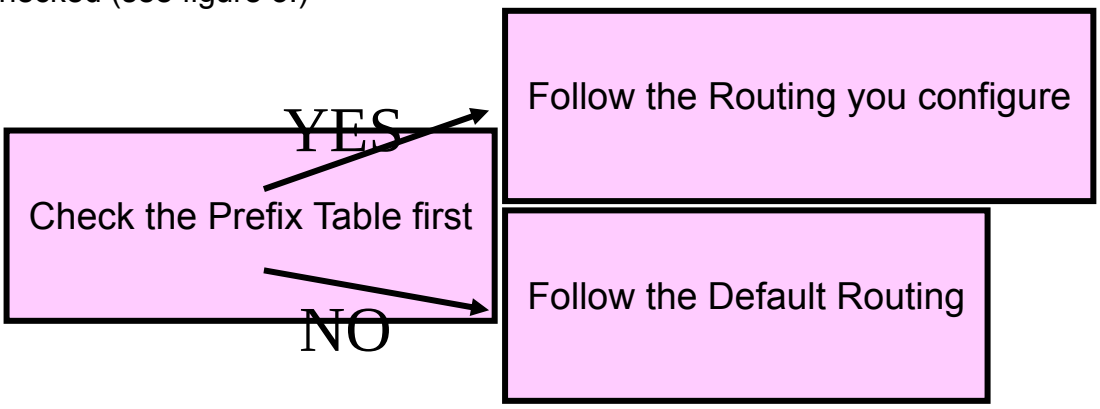


Figure 3.7: The Priority

There is an example about the configuration, please follow up these steps.

1 Press the Prefix Configuration button to enter the configuration table (see figure 3.6)

2 Enter the index number. Put the prefix numbers you will dial in the prefix table, enable (disable) the drop function and enter the numbers you want to insert (see figure 3.7)

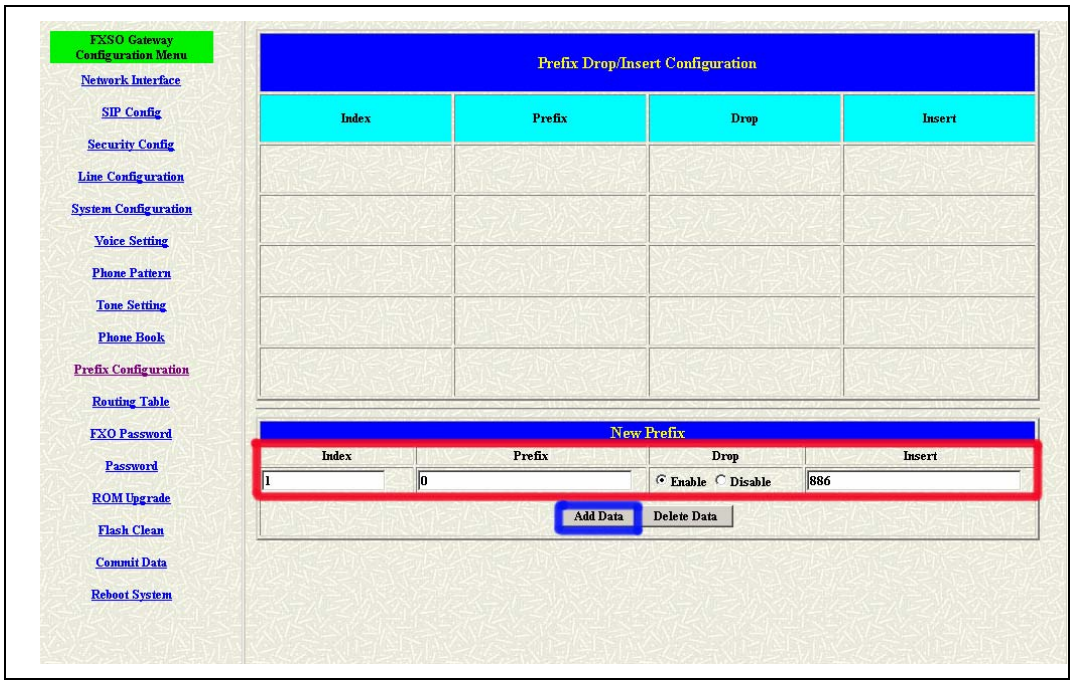


Figure 3.7: Configure the Prefix Table

The usage is as same as the drop, insert function of the Phone Book.

Input (Prefix)	Drop	Insert	Output
100	Disable	X	100
200	Disable	0	0200
300	Enable	X	X
400	Enable	500	500

3 Press the Prefix Configuration button to reload the configuration table (see figure 3.8)

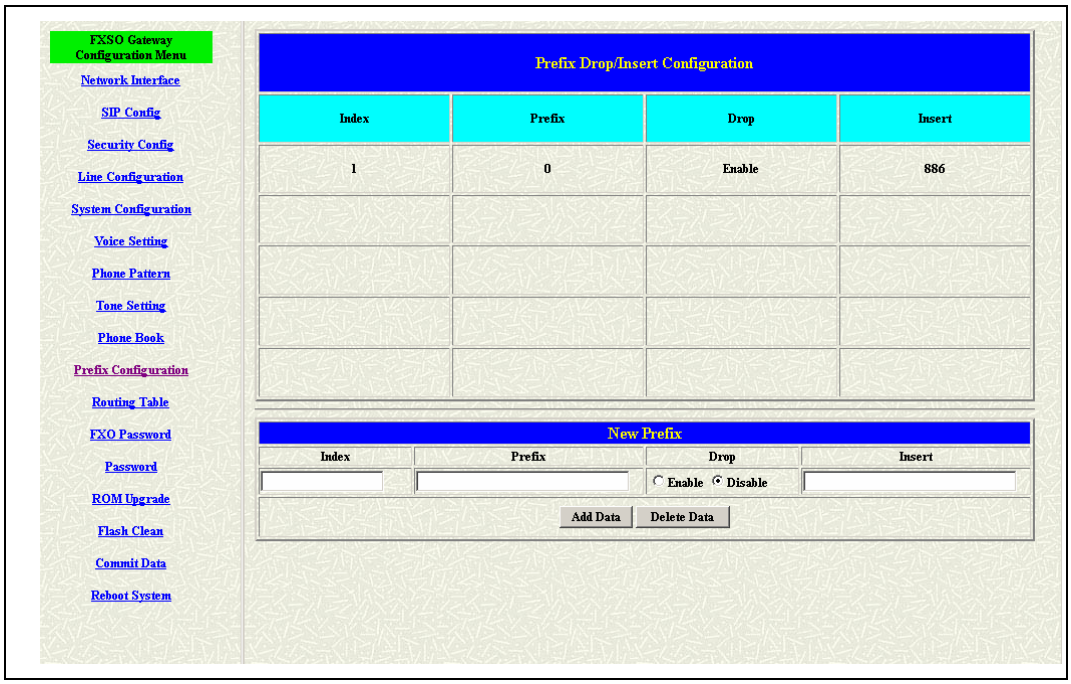


Figure 3.8: Show the added table

4 Please Commit it and Reboot the system if the configuration is finished.

3.7 Routing Table

Routing Table is a rule to define the destination of the calls you make. You could define the rules by the number you dial or by the ports. The Routing Table button will show you the configuration table (see figure 3.9).
In fact, there are three directions of the incoming calls (from IP, FXS and FXO side). The explanation of the default routing is as below:

The location with the incoming calls	The location with the destination	The explanation
IP (Default)	**FXS**	**The destination will be the FXS port when the calls from the IP side without any define rules.**
FXS (Default)	**IP**	**The destination will be the IP side when the calls from the FXS port without any define rules.**
FXO (Default)	**IP**	**The destination will be the IP side when the calls from the FXO port without any define rules.**

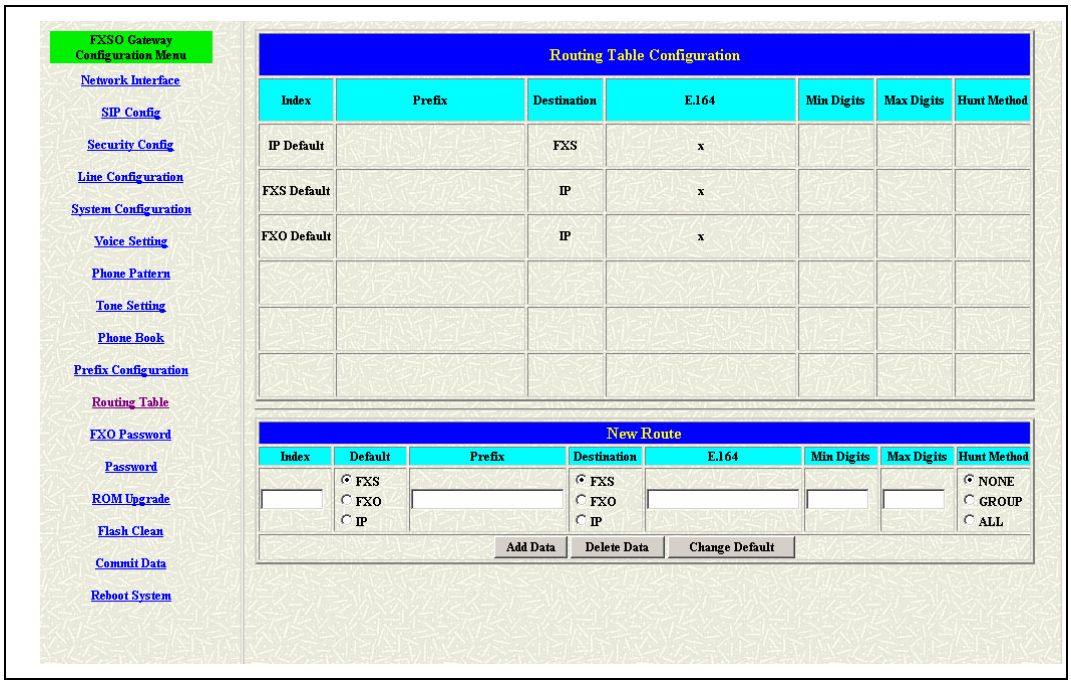


Figure 3.9: Routing Table Configuration

Change the default routing

Please follow up the steps if you want to change the default routing:

1 Pick up the side for the incoming calls and define the destination of this side. Press the Change Default to save the data. (see figure 3.10)

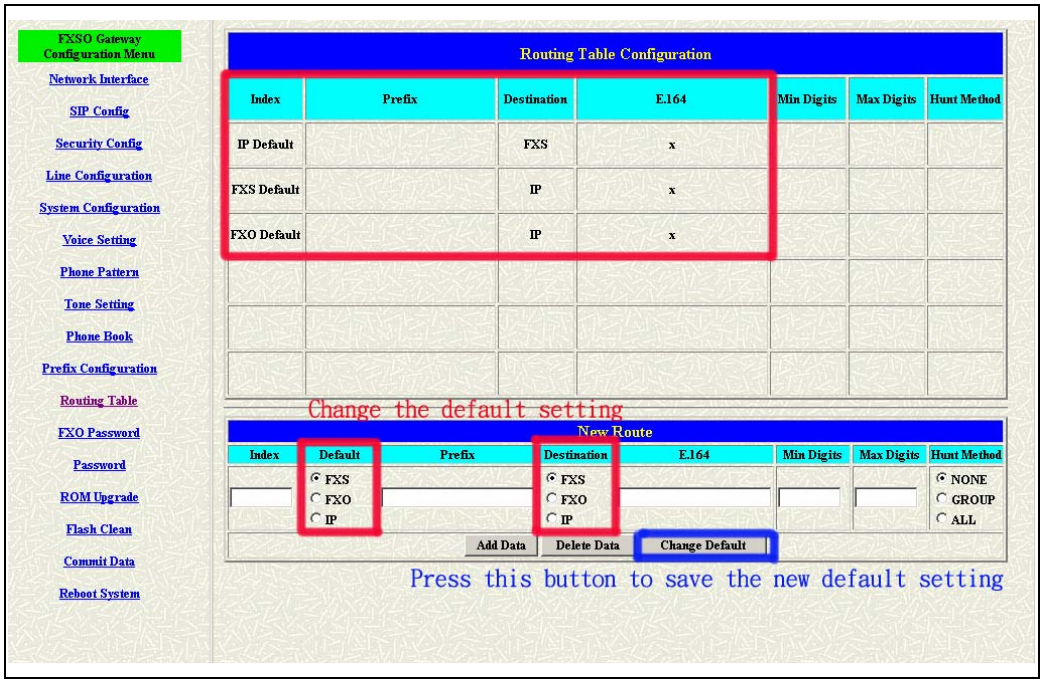


Figure 3.10: Change the Default Setting

2 The default setting is changed after you press the Change Default button.

Please press the Routing Table button again to show the new setting. (see figure 3.11)

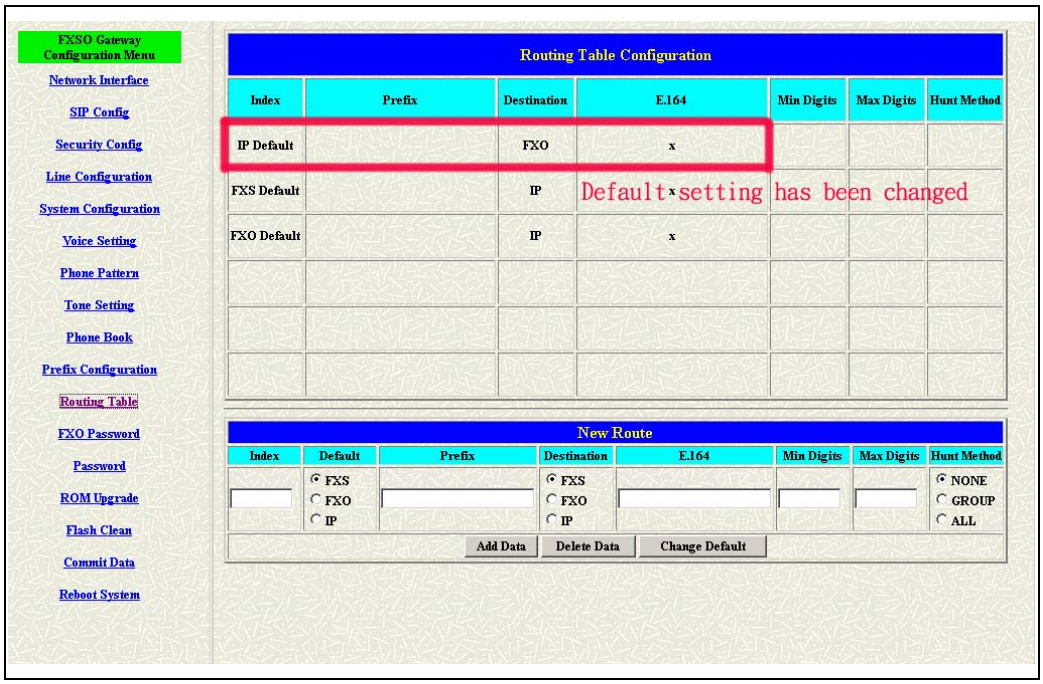


Figure 3.11: The Default Setting Changed

3 Please Commit it and Reboot the system if the configuration is finished.

Add a new Routing Table

1 The default setting is changed after you press the Change Default button.

Please press the Routing Table button again to show the new setting. (see figure 3.12)

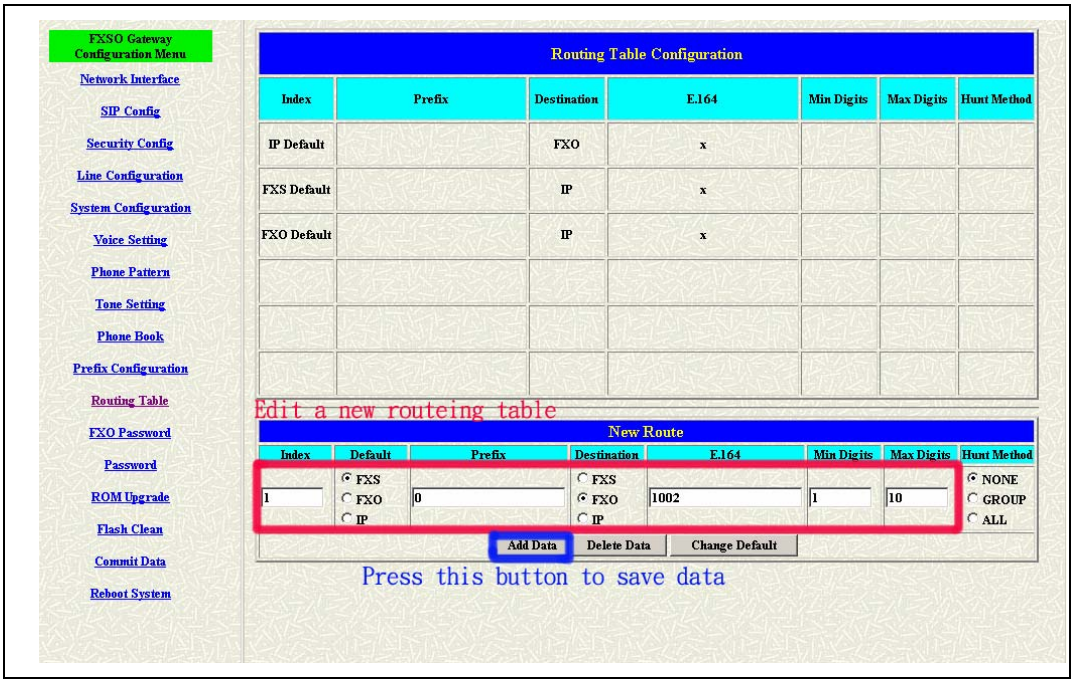


FIGURE 3.12: EDIT AND ADD A NEW ROUTING TABLE

- Index – Define the number of this data.
- Prefix – Define the number you dial. You could just define the first digit of the numbers
- Destination – Define the destination of this rule. There are three directions of the destination.
- E164 – Define a right E164 number of the destination you want.

For example: There are two FXO ports of the gateway (SP5014/S) and I want the first FXO port (1002 is the default E164 number) to be the destination. So the E164 number I have to define is 1002.

- Min Digits – The minima digits you dial.
- Max Digits – The maxima digits you dial.

The min and max digits are the range for the number you dial. For example: The min digits is 1 and max digits is 10. The call will follow

this routing if the number I dial is between 1 and 10 digits. If I dial over 10 digits, this call will follow the default routing.

- Hunt – Define the hunt group function.

None – Disable this function
Group – The call will search other ports to be the destination with the same group if the origin destination is busy.
All – The call will search other ports to be the destination with the same type if the origin destination is busy.

2 Press Add Data button to save the configuration and press the Routing Table button again to reload the configuration. (see figure 3.13)

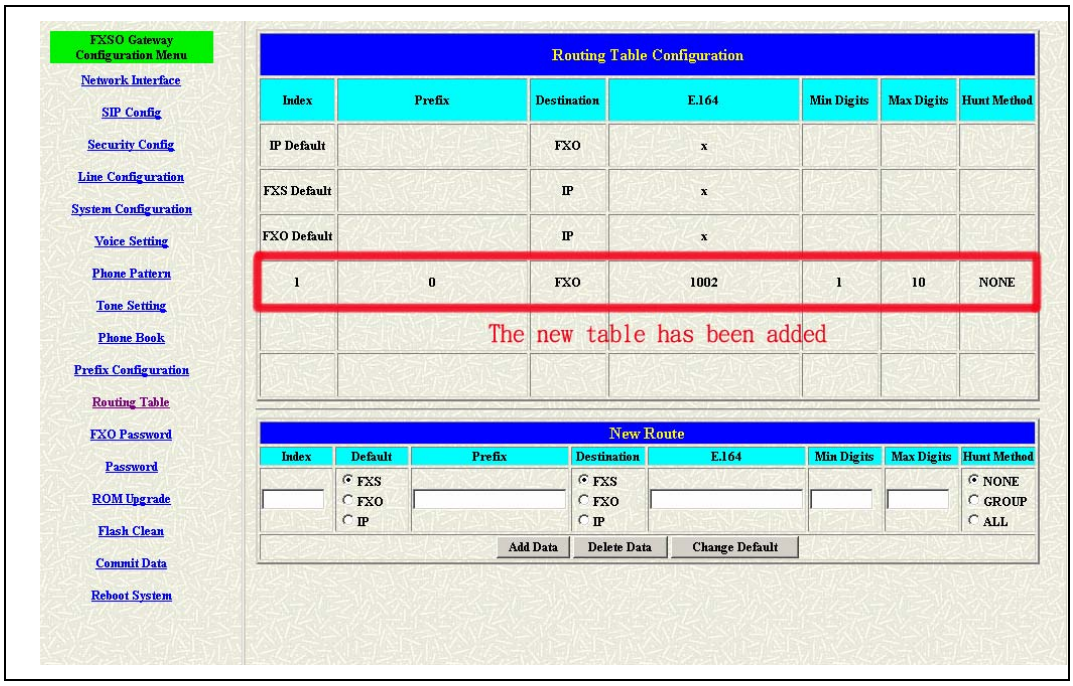


Figure 3.13: New Special Routing

When users dial 0 with the first digit of the numbers (from FXS side), or from FXO and IP side. And the numbers you dial is between 1 and 10 digits. If this call matches the rule, it will be transferred to the FXO port whose E164 number is 1002.

3 Please Commit it and Reboot the system if the configuration is finished.

3.8 FXO Password

You will get the IVR if you make calls from PSTN side. The IVR will ask you the password you set, and you could make other calls to IP side if the password you type is correct. Please press the FXO Password button to configure the password (see figure 3.14)

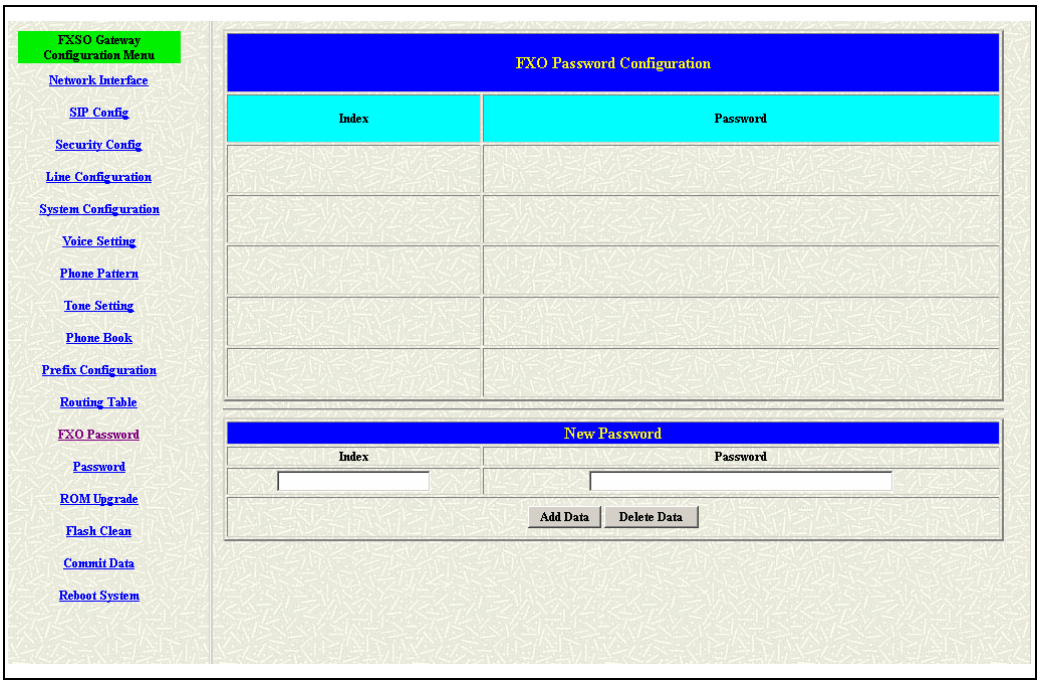


Figure 3.14: FXO Password

- Index – The number of this table.
- Password – The password you set.

This function is only for the calls from the PSTN side. It's not ready for the IP side as so far.

3.9 Password

There are two accounts for login to access or change the configurations. One is “root”, another is “administrator”. Users could define the password for these two login account. The account “root” could make all the configurations back to the default setting, but the account “administrator” couldn’t. This is the difference between these two accounts.
Users could define the password for the accounts in this page. (see figure 3.15)

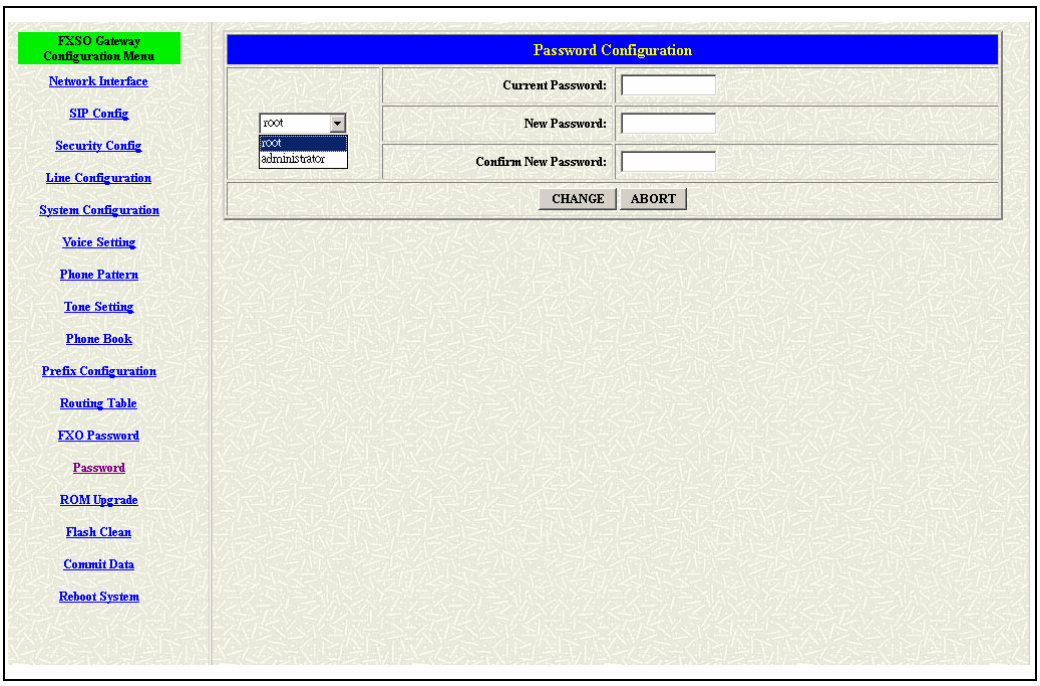


Figure 3.15: Password

- Account – The “root” could make all the configurations back to the default setting except the ip address and the password of the account. But the “administrator” couldn’t.
- Current Password – Enter the original password.
- New Password – Enter the new password, which you want.
- Confirm New Password – Enter the new password again.

Please remember the password you configure for the account. It will be more difficult to access it if you forgot the password.

3.10 Upgrade the Firmware

User could update the firmware just by the web configuration interface. There are two types for the upgrading procedure. One is using the TFTP server, another is using the FTP server. Please follow the step to update the gateway firmware version.

【Updating the firmware by the FTP server】

1 Pick up the “Rom Upgrade” button to enter the upgrading web page and switch to the FTP method. (see figure 3.16)

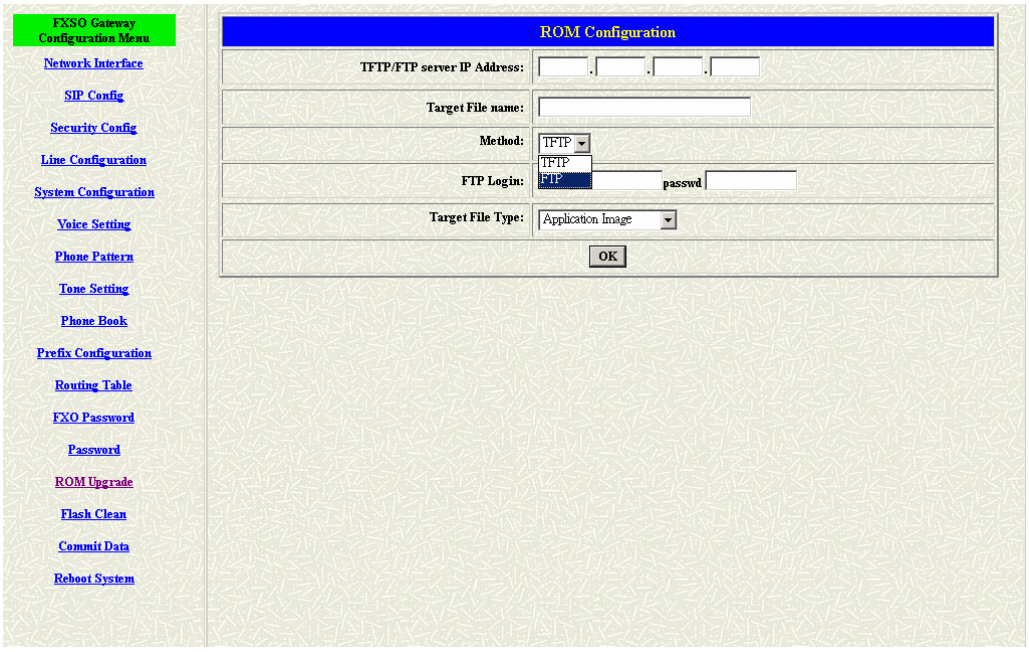


Figure 3.16: ROM Upgrade for FTP

2 Key in the IP address, the login name, password of your FTP server and the correct file name. (see figure 3.17)

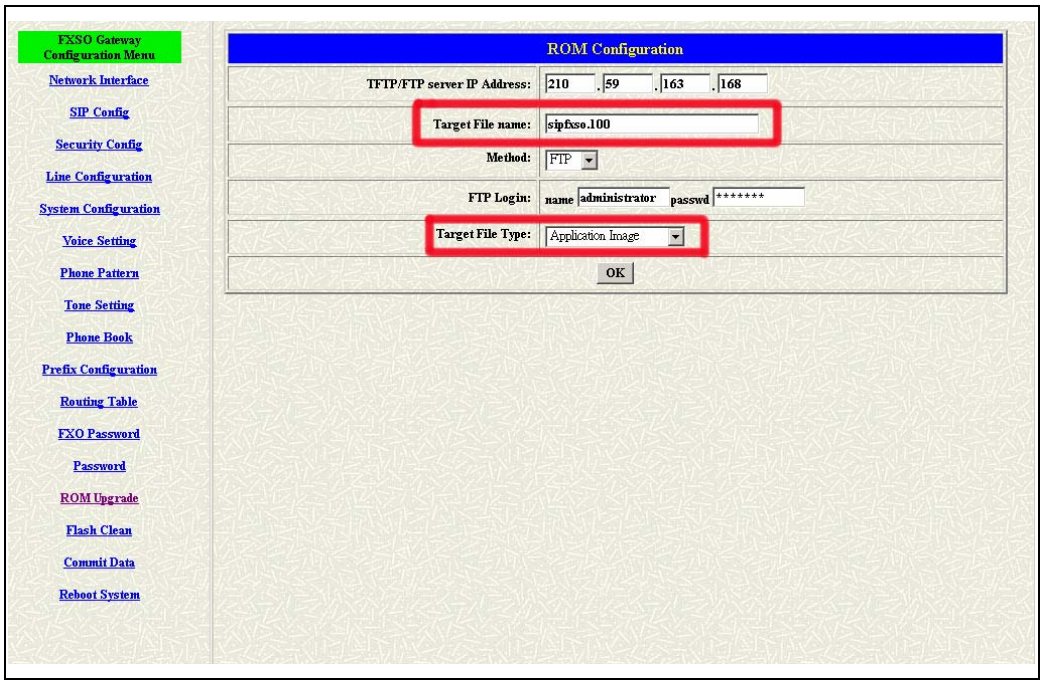


Figure 3.17: FTP information

Please pay more attentions about the red blank. The Target File Type has to be matched with the Target File name. Please put the correct info about the Target file in this table.

3 Press the OK button to execute the upgrade procedure.

4 Please press the “Reboot System” button to make it reboot.

【Updating the firmware by the TFTP server】

1 Downloading the TFTP program from our web site and install it first.

Execute the TFTP program before you want to use the TFTP upgrade method.

2 Pick up the “Rom Upgrade” button to enter the upgrading web page and

switch to the TFTP method. (see figure 3.18)

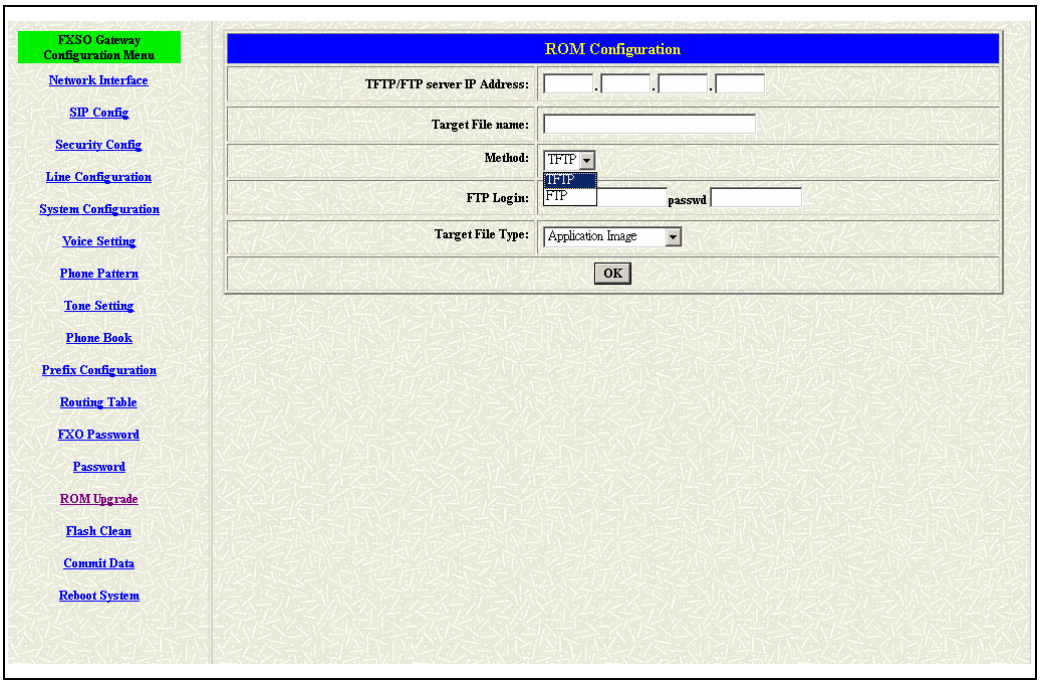


Figure 3.18: ROM Upgrade for TFTP

3 Key in the IP address of your TFTP server, pick up the file type for your upgrade file and the correct file name for upgrading. (see figure 3.19)

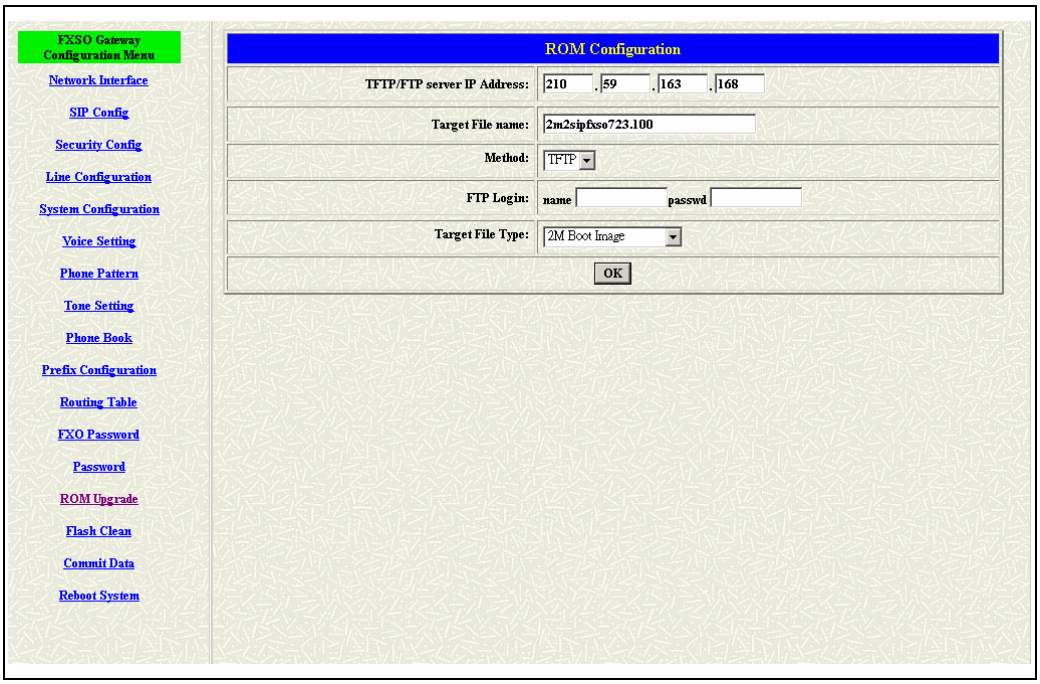


Figure 3.19 : TFTP information

4 Press the OK button to execute the upgrade procedure.

5 Please press the “Flash Clean” button when the procedure is finished.

6 After pressing the “Flash Clean” button, please press the “Reboot System” button to make it reboot.

4 Console Setup

4.1 Hyper Terminal Setting

A terminal emulator is needed when using RS-232 port to configure Gateway. There are kinds of terminal emulator software. Here, we use Microsoft HyperTerminal to depict how to set up terminal emulator:

1. Execute the *Hyper Terminal* program, and then the following windows will pop-up on the screen. (START – Program files – Accessories – Communication – Hyper Terminal)

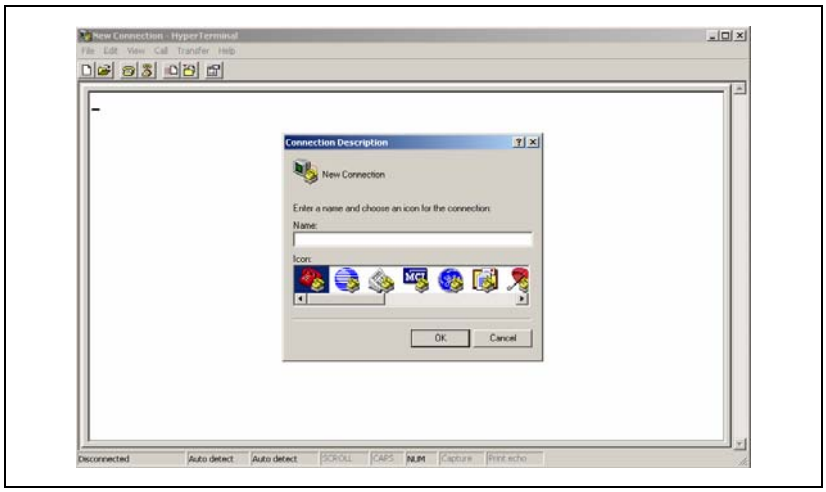


Figure 4.1: Hyper Terminal

2. Define a name such as ‘SP5012’ for this new connection.

Figure 4.2: Edit the name of the connection

3. After pressing OK button, the next window appear, and then choose ***COM1/2 Port***, which you are going to use.

Figure 4.3: Pick up the right interface to use

4. Configure the COM Port Properties as following:
   - Bits per second: 9600
   - Flow control: None

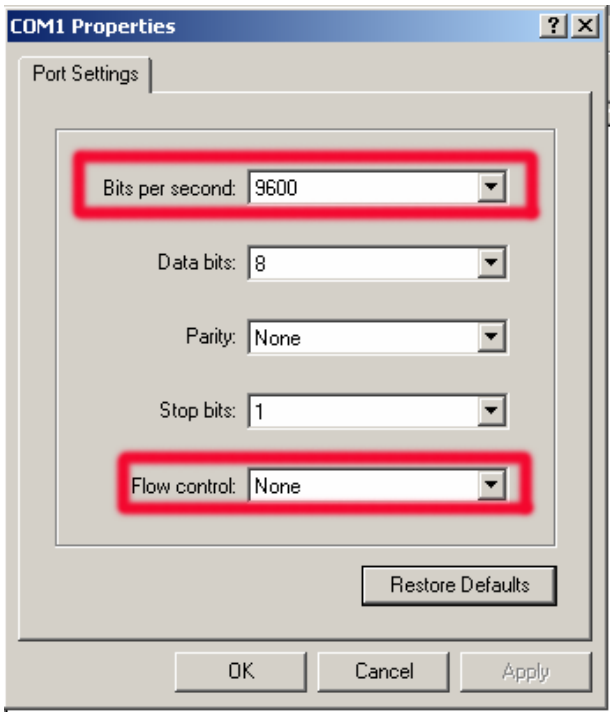


Figure 4.4: Configure the right Bps and control

5. Press ‘OK’ button, and then start to configure Gateway.

5 Command List

5.1 [help]

Type **help** or **man** or **?** to list all the available command.

usr/config$ help

help	help/man/? [command]
quit	quit/exit/close
debug	show debug message
reboot	reboot local machine
flash	clean configuration from flash rom
commit	commit flash rom data
ifaddr	Internet address manipulation
time	show current time
ping	test that a remote host is reachable
sysconf	System information manipulation
sip	SIP information manipulation
security	Security information manipulation
line	Line information manipulation
route	Routing information manipulation
prefix	Prefix drop/insert information manipulation
pbook	Phone book information manipulation
voice	Voice information manipulation
phone	Setup of call progress tones and ringing
tone	Setup of disconnect tone
fxopwd	Setup of FXO password
record	Record voice for greeting and ask pin code
pt	DSP payload type configuration and information
rom	ROM file update
passwd	Password setting information and configuration

usage: help [command]

5.2 [quit]

Type **quit** will quit the Gateway configuration mode and turn back to login prompt (in console mode) or disconnect (in TELNET mode).

```
usr/config$ quit
Disconnecting...
login:
```

Note: It is recommended that type the "**quit**" command before you leave the console. If so, Gateway will ask password again when next user connects to console port.

5.3 [debug]

Open debug message will show up specific information while Gateway is in operation. After executing the debug command, it should execute command **debug -open** as well. One example is demonstrated below.

```
usr/config$ debug -add fsm vp
usr/config$ debug -open
```

In this example, user open debug flags including fsm, vp.

Parameters Usage:

-status	Display the enabled debug flags.
-add	Add debug flag.
	-- fsm: sip related information
	-- vp : voice related information
-delete	Remove specified debug flag.
-open	Start to show debug messages.
-close	Stop showing debug messages.

5.4 [reboot]

After **commit** command, type **reboot** to reload Gateway in new configuration. The procedure is as below:

```
usr/config$ reboot

.Attached TCP/IP interface to cpm unit 0
Attaching interface lo0...done

Hardware auto detect...
Hardware Type : 1FXS + 1FXO
HTTPD initialized...
VoicePacketizermain comming
WorkMode : PROXY_MODE
incoming InitCallArray....REAL_MAXCALL=4
SIP stack was constructed successfully. Version - 2.2.1.8
Start registering to Proxy server

AC4804[0] is ok
successful 1 4
Initialize OSS libraries...OK!
VP v1.44 stack open successfully.

login:
```

5.5 [flash]

This command will clean the configuration stored in the flash ROM and reboot Gateway in factory default setting.

Parameter Usage:

-clean clean all the user defined values, and reboot Gateway in factory default mode.

Note: It is recommended that use "flash –clean" after application firmware id upgraded.

Warning:	Only user who login with **root** can execute this command. Configurations of IP address and accounts' passwords will be kept.

5.6 [commit]

Save changes after configuring Gateway.

---

usr/config$ commit

This may take a few seconds, please wait....
Commit to flash memory ok!
usr/config$

*Note: Users shall use **commit** to save modified value, or they will not be activated after system reboot*.

5.7 [ifaddr]

Configure and display Gateway network information.

usr/config$ ifaddr

LAN information and configuration
Usage:
ifaddr [-print]\|[-dhcp used]\|[-sntp mode [server][-cmcenter ipaddress]]
ifaddr [-ip ipaddress] [-mask subnetmask] [-gate defaultgateway]

-print Display LAN information and configuration.
-ip Specify ip address.
-mask Set Internet subnet mask.
-gate Specify default gateway ip address
-ipmode Set ip client service (0=Fix ip, 1=DHCP, 2=PPPoE).
-sntp Set SNTP server mode and specify IP address.
-timezone Set local timezone.
-ipsharing Specify usage of an IP sharing device and specify IP

address.

-ipchange Replace IP address if the shared IP is changed.

-cmcenter Specify IP address of management center.

Note:

SNTP mode (0=no update, 1=specify server IP, 2=broadcast mode).

Example:

ifaddr -ip 210.59.163.202 -mask 255.255.255.0 -gate 210.59.163.254

ifaddr -ipmode 1

ifaddr -sntp 1 210.59.163.254

ifaddr -ipsharing 1 210.59.163.254

ifaddr –ipchange 1

usr/config$

Parameters Usage:

-print	print out current [ifaddr] settings and status
-ip	assign IP address for Gateway
-mask	assign internet subnet mask
-gate	assign IP default gateway
-mode	Switch the network type (0 = Static IP; 1 = DHCP mode 2 = PPPoE mode)
-sntp	Simple Network Time Protocol (1 = ON; 0 = OFF) When SNTP function is activated, users have to specify a SNTP server as network time source. An example is demonstrated below:
-timezone	set local time zone according to GMT

---

usr/config$ ifaddr -sntp 1 10.1.1.1

10.1.1.1 stands for SNTP server’s IP address.

-ipsharing To specify a global fixed IP address, user can add this IP address in the command.

usr/config$ ifaddr -ipsharing 1 210.11.22.33

If the IP address is not a fixed one, the dedicated IP address is not necessary in the command. However,

dynamic IP Address is not working in Peer-to-Peer mode.

---

usr/config$ ifaddr -ipsharing 1

-ipchange This is for a special function of IP-Sharing. Some devices could support different ip address from WAN and DMZ port. This function has to be enable if the ip address of DMZ port is different from the WAN port.

---

usr/config$ ifaddr -ipchange 1

-cmcenter Specify a management center IP Address. Micronet will provide a management center program to do centralized control for all devices.

5.8 [time]

When SNTP function of Gateway is enabled and SNTP server can be found as well, type **time** command to show current network time.

---

usr/config$ time
Current time is THU JAN 01 05:29:23 1970

5.9 [ping]

Use **ping** to test whether a specific IP is reachable or not.
For example: if 192.168.1.2 is not existing while 192.168.1.254 exists. Users will have the following results:

usr/config$ ping 192.168.1.2
no answer from 192.168.1.2
usr/config$ ping 192.168.1.254

PING 192.168.1.254: 56 data bytes
64 bytes from 192.168.1.254: icmp_seq=0. time=5. ms
64 bytes from 192.168.1.254: icmp_seq=1. time=0. ms
64 bytes from 192.168.1.254: icmp_seq=2. time=0. ms
64 bytes from 192.168.1.254: icmp_seq=3. time=0. ms
----192.168.1.254 PING Statistics----
4 packets transmitted, 4 packets received, 0% packet loss
round-trip (ms) min/avg/max = 0/1/5
usr/config$

5.10 [sysconf]

This command displays system information and configurations.

---

usr/config$ sysconf

System information and configuration
Usage:
sysconf [-idtime digit][-keypad dtmf][-prefixsw digit]
[-prefixdisab digit][-usrdefprefix digits]
[-codec digit][-localrbt digit][-forwardtime digit]
[-hwtype digit][-gwprefix digit][-ring on_time off_time]
[-fxotype digit]
sysconf -print

-print Display system overall information and configuration.
-idtime Inter-Digits time.(1~10 sec)
-forwardtime Forward time for FXS line if no answer.(5~65535 sec)
-keypad Select DTMF type: 0=In-band,
1=RFC2833.
-hwtype Hardware type.(Auto:0 / 1FXS+1FXO:1 /
2FXS+2FXO:2)
-ring The ring time for ring detection.(Uint:ms)
-rba The number of ring times before answer.(1~5)

-eod End of dial.(Enable: 1 / Disable : 0)

Example: sysconf -ring 500
usr/config$

Parameters Usage:

-print print out all current settings

-idtime set the duration (in second) of two pressed digits in dial mode as timed out. If after the duration user hasn't pressed next number, it will dial out all number pressed. (1-10 seconds)

-forwardtime set forward time (5-65535 seconds) for FXS Line. If callee hasn't answered the call in this time, call will be forward to assigned number in [line] command. (Please refer to ***[line]*** command for forward setting)

-keypad DTMF replay type. When value is "0", Gateway will transfer DTMF signal via In-Band type, "1" via RFC2833 type. Users can adjust the value according to various applications.

1. *Number (instead of Line number of FXO Line)+ PSTN number to make a call to PSTN side connected with FXO Line.*
2. *After gateway-prefix-drop function is enabled, user must remember to re-configure line number of FXS Line, because line number of FXS Line must remove prefix number. For example, origin line number of FXS line is 1001, prefix is 100, since prefix number will be drop, once gateway has incoming call 1001, after drop gateway prefix 100, it will search line number "1". So line number must be set as "1".*

-hwtype application rom file of 37 series are the same no matter how many ports is the module, so after user downloads the application rom file, user can select which hardware type is . "0" means gateway will automatically detect the hardware type, "1" means the hardware type is 1FXS+1FXO, "2" means the hardware type is 2FXS+2FXO.

Note:

The default value is to auto detect hardware type. Usually it is not necessary to change this setting. Please make sure about your Hardware

Type, Gateway may be not functional if set wrong hardware type.

-ring ring time for ring detection (in ms). When Gateway has incoming call from PSTN side to FXO port, Gateway will determine it is a ring but not noise only if it is longer than this ring time.

Note:
In Taiwan the ring time of PSTN usually is 1000ms, so if user set ring time longer that 1000ms, FXO port may not be able to pick up the call from PSTN side.

-rba When the calls from the PSTN side, FXO port will off hook if the ring time is matched with this number.

-eod It will transfer the DTMF in "#" if users disable the end of dial function. Users have to press the key pad in "#" if the end of dial function is enable.

5.11 [sip]

This command is for sip configuration related parameters.

---

usr/config$ sip
sip stack information and configuration
Usage:

-print	*Display SIP stack information and configuration.*
-mode	*Configure as Proxy mode or Peer-to-Peer mode.*
-px	*Proxy server address. (Proxy IPv4 address or Proxy dns name)*
-domain	*Second domain name in the URL (if domain name is not used, specify as null)*
-prefix	*Specify prefix string, use it when the UserID contains alphabets*
-line1	*Line 1(TEL 1) SIP number.*
-line2	*Line 2(LINE 1) SIP number.*
-line3	*Line 3(TEL 2) SIP number.*
-line4	*Line 4(LINE 2) SIP number.*

-expire *The relative time after which the message expires (0~65535).*

-port *SIP local UDP port number (5060~5070). Default : 5060*

-rtp *RTP port number (2326~65532). Default : 16384*

Example:

sip -px 210.59.163.171 -line1 70 -line2 71 -line3 72 -line4 73

usr/config$

Parameters Usage:

-print print current h323 related settings

-mode alternatives for proxy or peer-to-peer mode (1=proxy mode; 0=peer-to-peer mode). If users select proxy mode, a valid proxy is needed when Gateway is in operation.

---

usr/config$ sip –mode 0 (peer to peer mode)

-px to assign the ip address of the proxy when Gateway is in proxy mode.

-domain to assign the domain name of the proxy when it is needed.

-prefix this will be prefix the alphabets before the sip line number.

-line1 assign FXS TEL1 number.

-line2 assign FXO Line1 number.

-line3 assign FXS TEL2 number.

-line4 assign FXO Line2 number.

Note:

User can also set "x" in line number to disable the port. If the port is disabled, it can only receive calls but not calling out.

Note:

1. This is for SP5014/S, for SP5012/S, there are only line1 and line2 command.
2. No matter in Proxy or P2P mode, user only needs to dial line number to reach local port. For example, in P2P mode, user wants to dial from FXS TEL1 to FXO Line1, only need to dial number of line2.

-expire It just like the TTL function in H323, the gateway will make sure the registration is success or not for a period times.
-port define the local sip port for this gateway.
-rtp to allocate RTP port range—NOT RECOMMENDED. This may be used when RTP port range conflicts with Firewall policy. (each port of Gateway use 2 RTP ports)

Note: From –rtp to –conneto commands are for advanced users, please do not change the default settings if not necessary.

5.12 [security]

This is the authentication for the SIP account.

---

usr/config$ line

Security information and configuration
Usage:
security [-name username] [-password password]
security -print

-print Display system account information and configuration.
-line Specify which line number you want to set the account.
-name Specify user name.
-password Specify password.
Example:
security -line 1 -name test -password 12345

Parameter Usages:
-print print out all current settings of security.
-line the line number, which you want to define the security info
-name the name is as same as the SIP number.
-password the password for the authentication if it is the necessary for the proxy.

5.13 [line]

This command is for configure each line parameters of Gateway.

---

usr/config$ line

Gateway line information and configuration
Usage:
line -config number [hunt number][hotline number][forward number]
line -print Gateway line information.
hunt Hunting group.
hotline Hot line configuration.
forward No answer forward for FXS line.
Example:
line -config 1 hunt 1 hotline 1003 forward 1002

usr/config$

Parameter Usages:

-print print out all current settings of line

-config determine which line to configure

-hunt set hunting group flag of each line. User can assign different hunt group number represent different hunt group. For example, if user assigns FXS TEL1 as hunt group 1, and FXS TEL2 as hunt group 2, they will be determined as 2 different groups. On the other hand, if user assigns FXS TEL1 as hunt group 1, and FXS TEL2 as hunt group 1 too, when having incoming call to FXS TEL1, which is busy, this call will be route to FXS Line2.

Note: *FXO Lines and FXS TELs are treated as 2 different groups, so even they are in the same hunt group, call will only be routed to the same FXS or FXO Lines.*

-hotline set hotline table. The Hotline Mode is applied in limited two channels. User just picks up the phone set of one FXS TEL or calls in one FXO line, and gateway will automatically dial out a phone number. In the other hand, user will hear ring back tone or dial tone immediately depended on configurations of destination device. **Note: This function can both work in**

Proxy or P2P mode.

(1) Call out from FXS Line

Proxy Mode Usage:

Set gateway under proxy mode.

Create a Hotline table with "***line*** " command.

---

usr/config$ line –config 1 hotline 1001

In this example means: if user picks up phone set of FXS Line1, gateway will automatically dial out "1001".

P2P Mode Usage:

Set gateway under P2P mode.

Create phone book table with "***pbook*** " command.

Create a Hotline table with "***line*** " command.

---

usr/config$ pbook –add name micronet ip 10.1.1.1 e164 1001

usr/config$ line –config 1 hotline 1001

In this example means: if user picks up phone set of FXS Line1, gateway will automatically dial out IP address of "1001".

(2) Call out from FXO Line

Proxy Mode Usage:

Set gateway under proxy mode.

Create a Hotline table with "***line*** " command.

---

usr/config$ line –config 2 hotline 1001

In this example means: if user calls in FXO Line1, gateway will automatically dial out "1001".

P2P Mode Usage:

Set gateway under P2P mode.

Create phone book table with "***pbook*** " command.

Create a Hotline table with "***line*** " command.

---

usr/config$ pbook –add name micronet ip 10.1.1.1 e164 1001
usr/config$ line –config 2 hotline 1001

In this example means: if user calls in FXO Line1, gateway will automatically dial out IP address of “1001”.

-forward set no answer forward table for FXS Lines.
Only **FXS** Lines provides No Answer Forward function. For call forward function, it can works under Proxy or P2P mode.
(1) Proxy Mode Usage:

---

usr/config$ line –config 1 forward 1002

In this example means: if user calls in FXS Line and hasn’t been answered in forward time (please refer to ***[sysconf -forwardtime***] command), gateway will automatically forward this call to phone number “1002”.

(2) P2P Mode Usage:

---

usr/config$ line –config 1 forward 1002

In this example means: if user calls in FXS Line1 when Line1 is busy, gateway will automatically forward this call to IP address of “1002” in Phone book.

5.14 [route]

This command is to set routing table for Gateway.

usr/config$ route
Routing table information and configuration
Usage:
route -add [prefix number][dst number][e164 number]
[min number][max number][hunt number]
route -delete index
route -modify index [prefix number][dst number][e164 number]

[min number][max number][hunt number]

route -ip [dst number][SIP number]

route -fxs [dst number][SIP number]

route -fxo [dst number][SIP number]

route -print Routing table information.

prefix The prefix of dialed number.

dst Destination port(FXS:0/FXO:1/IP:2).

e164 Destination e164 number(when destination is FXS or FXO).

min Min digits.(0 ~ 255)

max Max digits.(0 ~ 255)

hunt Hunt method for busy forward(NONE:0 / GROUP:1 / ALL:2)

Example:

route -add prefix 100 dst 0 e164 1001 min 1 max 3 hunt 1

route -ip dst 0 e164 1001

route -fxs dst 2

route -fxo dst 0 e164 x

route -modify 1 prefix 100 dst 0 e164 1001 min 1 max 3 hunt 1

route -delete 1

usr/config$

Parameter Usages:

-print print out all routing table information

-add add a routing rule in routing table. User can add less than 50 rules. (***route –add prefix “prefix number” dst “destination port type” e164 “SIP number of port” min “minimum digits needed” max “maximum digits can’t be exceeded”***)

-delete delete a routing rule in routing table (***route –delete “index of routing rule”***)

-modify modify a routing rule in routing table. (***route –modify “index of routing rule” prefix “prefix number” dst “destination port type” e164 “SIP number of port” min “minimum digits needed” max “maximum digits can’t be exceeded”***)

-ip create routing table for incoming call from IP side. (***route –ip dst “destination port type” e164 “SIP number of port”***)

-fxs	create routing table for incoming call from FXS TELs. (***route –fxs dst "destination port type" e164 "SIP number of port"***)
-fxo	create routing table for incoming call from FXO Lines. (***route –fxo dst "destination port type" e164 "SIPnumber of port"***)
prefix	prefix of dialed number
dst	destination port, 0 means FXS TELs, 1 means FXO Lines, 2 means IP side, x means no determinate number.
e164	destination SIP number. This only need to be set when routed port is FXS TELs or FXO Lines to determine which port will this call be routed to.
min	minimum digits needed.
max	maximum digits needed.
hunt	set hunt method for busy forward. 0 means no hunting, 1 means hunting method follows the rule of ***[line]***, 2 means hunting method is to hunt between all ports in the same type, for example, destination port is FXS TEL will hunt in all FXS TELs, destination port is FXO Lines will hunt in all FXO Lines.

Usage Example:

1. ***route –add prefix 100 dst 0 e164 1001 min 1 max 3 hunt 1***
   This command means if gateway has incoming call's prefix number is 100, and total digits is between 1 to 3, this call will be routed to FXS TEL 1001, and if TEL 1001 is busy, call will be routed to another FXS TEL.
2. ***route –ip dst 1 e164 1002***
   This command means incoming call from IP side will be routed to FXO Line of number 1002.
3. ***route –fxs dst 1 e164 1002***
   This command means incoming call from FXS TELs will be routed to FXO Line of number 1002.
4. ***route –fxo dst 2***
   This command means incoming call from FXO Lines will be routed to IP side.

Note:

(1) When destination is IP side, SIP number doesn't need to

determine. (Ex. route –fxs dst 2)

(2) If user doesn't want to determine a specific port to route, SIP number must set as "x". (Ex. route –ip dst 1 e164 x)

(3) Default value: Incoming call from FXS and FXO ports will be forward to IP side directly.

5.15 [prefix]

This command is for make rules for drop or insert prefix digits.

---

usr/config$ prefix

Prefix drop/insert information and configuration
Usage:
prefix -add [prefix number][drop number][insert digits]
prefix -delete index
prefix -modify index [prefix number][drop number][insert number]
prefix -print Prefix drop/insert information.
prefix The prefix of dialed number.
drop Drop prefix(Enable:1/Disable:0).
insert Insert digits.
Example:
prefix -add prefix 100 drop 1 insert 2000
prefix -add prefix 100 drop 1
prefix -add prefix 100 drop 0 insert 200
prefix -delete 1
prefix -modify 1 prefix 100 drop 0 insert 300

usr/config$

Parameter Usages:

-add add a rule to drop or insert prefix digits of incoming call. (***prefix –add prefix "prefix number" drop 0/1 insert "insert number"***)

-delete delete a rule to drop or insert prefix digits of incoming call.

(***prefix –delete prefix "prefix number"***)

-modify modify a rule to drop or insert prefix digits of incoming call.
(***prefix –modify prefix "prefix number" drop 0/1 insert "insert number"***)

prefix set which prefix number to implement prefix rule.

drop enable or disable drop function. If this function is enabled, Gateway will drop prefix number on incoming call.

insert set which digit to insert on incoming call.

5.16 [pbook]

Phone Book function allows users to define their own numbers, which mapping to real IP address. It is effective only in peer-to-peer mode. When adding a record to Phone Book, users also have to reboot the machine, and the record will be effective immediately.

---

usr/config$ pbook

Phone book information and configuration
Usage:
pbook [-add [name string][e164 number][ip address]
[port number][drop digit][insert number]]
[-modify number [name string][e164 number][ip address]
[port number][drop digit][insert number]]
[-delete number]
pbook -print

-print Display phone book information and configuration.
-add Add new phone book record)
-delete Delete phone book record
-modify Modify phone book record.
name : 1 ~ 10 characters.
e164 : 1 ~ 10 digits.
ip : IP address.
port : 1024 ~ 65535.
drop : 0:Disable/1:Enable.
insert : 1 ~ 10 digits.

Example:

pbook -add name test e164 1234 ip 192.168.1.10 drop 1 insert 5678

pbook -delete 1

pbook -modify 1 name test e164 5678 ip 192.168.1.10 drop 0

usr/config$

Parameter Usages:

-print	print out current contents of Phone Book. (***pbook -print***) Users can also add *index number*, from 1 to 100, to the parameter to show specific phone number. (Ex. ***pbook –print 1***)
	Note: <index number> means the sequence number in phone book. If users do request a specific index number in phone book, Gateway will give each record an automatic sequence number as index.
-add	add a new record to phone book. When adding a record, users have to specify ***name***, ***ip***, and ***e164*** number to complete the command.
name	name to represent callee.
e164	The SIP number for mapping with IP address of called
ip	ip address of called
drop	drop e.164 number when dial out. 0 means to keep e.164 number, 1 means to drop e.164 number when dialing out.
insert	insert digits.(1~10 digits)
-delete	delete a specific record. “pbook –delete 3” means delete **index 3** record.
-modify	modify an existing record. When using this command, users have to specify the record’s index number, and then make the change.

PhoneBook Rules:

The SIP number defined in phone book will fully carry to destination. It is not just a representative number for destination’s IP Address. In other words, user dial this number to reach the destination, destination will receive the number and find out if it is matched to itself, including Line number in some particular device.

5.17 [voice]

The voice command is associated with the audio setting information. There are four voice codecs supported by Gateway.

---

usr/config$ voice

Voice codec setting information and configuration
Usage:
voice [-send [G723 ms] [G711A ms] [G711U ms] [G729 ms] [G729A ms] [G729B ms] [G729AB ms]]
[-volume [voice level] [input level] [dtmf level]]
[-nscng [G711U used1] [G711A used2] [G723 used3]]
[-echo used] [-mindelay t1] [-maxdelay t2] [-optfactor f]
voice -print
voice -priority [G723] [G711A] [G711U] [G729] [G729A] [G729B] [G729AB]

-print Display voice codec information and configuration.
-send Specify sending packet size.
G.723 (30/60 ms)
G.711A (20/40/60 ms)
G.711U (20/40/60 ms)
G.729 (20/40/60 ms)
G.729A (20/40/60 ms)
G.729B (20/40/60 ms)
G.729AB (20/40/60 ms)
-priority Priority preference of installed codecs.
G.723
G.711A
G.711U
G.729
G.729A
G.729B
G.729AB
-volume Specify the following levels:

voice volume (0~63, default: 29,28),
input gain (0~63, default: 26),
dtmf volume (0~31, default: 23),

-nscng No sound compression and CNG. (G.723.1 only, On=1, Off=0).

-echo Setting of echo canceller. (On=1, Off=0, per port basis).

-mindelay Setting of jitter buffer min delay. (0~150, default: 90).

-maxdelay Setting of jitter buffer max delay. (0~150, default: 150).

Example:

voice -send g723 60 g711a 60 g711u 60 g729 60 g729a 60 g729b 60 g729ab 60

voice -volume voice 20 input 32 dtmf 27

voice -echo 1 1

usr/config$

Parameters Usage:

-print print current voice information and configurations.

-send define packet size for each codec. 20/40/60ms means to send a voice packet per 20/40/60 milliseconds. The smaller the packet size, the shorter the delay time. If network is in good condition, smaller sending packet size is recommended. In this parameter, 20/40/60ms is applicable to G.711u/a law, and G.729/G.729A/G.729B/G.729AB codec, while 30/60ms is applicable to G.723.1 codec.

-priority codec priority while negotiating with other h323 device. This parameter determines the listed sequence in h.245 TCS message. The codec listed in left side has the highest priority when both parties determining final codec. User can also select the particular codec without others.

---

usr/config$ voice –priority g723 (only select this codec)

usr/config$ voice –priority g723 g729 g711u g711a (select four codecs, and g723 is the first choice)

Note:

(1) For SP5014/S there are 2 version of Application rom, please check out the version of Application rom (rom –print). If the version is 2sipfxso729.100, SP5014/S doesn't have the codec

G.723.1. If the version is 2sipfxso723.100, SP5014/S doesn't have the codec G.729 series.

(2) For SP5012/S, the Application rom has the only one version which is named sipfxso.100 provide all codec.

-volume There are three adjustable value. **voice volume** stands for volume, which can be heard from Gateway side; **input gain** stands for volume, which the opposite party hears; **dtmf** volume stands for DTMF volume/level, which sends to its own Line.

Note: level of volume is too high or too low may be result in bad performance while connecting to each other.

-nscng silence suppression and comfort noise generation setting (1 = ON; 0 = OFF). It is applicable to G.723 codec only. An example is demonstrated below:

---

usr/config$ voice -nscng g723 1

-echo activate each canceller (1 = ON; 0 = OFF).
-mindelay the minimum jitter buffer size. (Default value= 90 ms)
-maxdelay the maximum jitter buffer size. (Default value= 150 ms)

---

usr/config$ voice –mindelay 90 –maxdelay 150

Note: be sure to know well the application before you change **voice** parameters because this might cause incompatibility.

5.18 [phone]

Gateway's progress tone is configurable. Default tone value is set according to U.S. tone specification. Users may adjust the values according to their own country's tone specification or users-defined tone specification.

---

usr/config$ phone

Phone ringing , ringback tone , busy tone , dial tone setting and notes
Usage:

phone [-ring [freq][ringON][ringOFF][ringLevel]]
[-rbt [freqHi][freqLo][freqHiLev][freqLoLev]
[Tone1ON][Tone1OFF][Tone2ON][Tone2OFF]]
[-bt [freqHi][freqLo][freqHiLev][freqLoLev]
[Tone1ON][Tone1OFF][Tone2ON][Tone2OFF]]
[-dt [freqHi][freqLo][freqHiLev][freqLoLev]
[Tone1ON][Tone1OFF][Tone2ON][Tone2OFF]]
[-flash [freqLo][freqHi]]
phone [-print [ring]\[rbt]\[bt]\[dt]\[flash]]

-print Display phone ringing/tone configuration.
ring : ringing
rbt : ringback tone
bt : busy tone
dt : dial tone
flash: flash tone
-ring ringing configuration set.
-rbt ringback tone configuration set.
-bt busy tone configuration set.
-dt dial tone configuration set.
-flash flash configuration set.

Note:
ringing frequency : 15 ~ 100 (Unit : Hz)
ringing ring ON/OFF : 0 ~ 8000 (Unit : ms)
ringing level : 0 ~ 94 (Unit : V)
tone frequency : 0 ~ 65535 (Unit : Hz)
tone freqLevel : 0 ~ 65535 (Unit : mVrms)
tone Tone ON/OFF : 0 ~ 8000 (Unit : ms)
Example:
phone -print rbt
phone -ring 20 2000 4000 94

---

phone -rbt 480 440 8 8 2000 4000 2000 4000
phone -bt 620 480 8 8 500 500 500 500
phone -dt 440 350 8 8 500 1023 1023 1023
phone -flash 100 300

usr/config$

Parameters Usage:

-print print current call progress tone configurations (**ring** – ring tone, **rbt** – ring back tone, **bt** – busy tone, **dt** – dial tone, flash – flash). This parameter should be accompanied with tone type. For example:

---

usr/config$ phone -print rbt

Phone ring back tone parameter
Ringback Tone frequency high : 480
Ringback Tone frequency low : 440
Ringback Tone frequency high level : 13
Ringback Tone frequency low level : 13
Ringback Tone tone1 on : 100
Ringback Tone tone1 off : 200
Ringback Tone tone2 on : 1023
Ringback Tone tone2 off : 1023
usr/config$

Note:
For tone simulation, Gateway adopts dual frequencies as traditional telephone does. If users want to have their own call progress tone, they can change the value of tones. High and Low frequency/level/cadence can be configured respectively.

-ring to set RING tone value.
The played tone type, when Gateway is receiving a call.
-rbt to set RingBackTone value
The played tone type, when Gateway receives a Q.931 Alerting message. In condition that Gateway is the originate side.
-bt to set BusyTone value.
The played tone type, when destination is busy.

-dt to set DialTone value.
The played tone type, when hook off a phone set of workable Gateway.
-flash set the detective flash range in ms, for example, 300-500 ms.

5.19 [tone]

This command is basically for FXO ports.

---

usr/config$ tone

Disconnect tone and remote ring back tone configuration
Usage:

tone [num][freqHi][freqLo][freqHiLev][freqLoLev]
[Tone1ON][Tone1OFF][Tone2ON][Tone2OFF]]
tone -print Display disconnect tone configuration.
[num] Tone index(1~4:Disconnect tone / 5~8:Remote ring back tone).
Example:
tone -print
tone 1 620 480 8 8 50 50 1023 1023

usr/config$

Parameter Usages:
-print show all tone configuration
[num] tone index. 1~4 is disconnect tone, 5~8 is remote ring back tone.
For FXO ports Gateway must detect disconnect tone to determine when to disconnect the call, so user must set disconnect tone of PBX or PSTN network connected to FXO ports.
When making a call from FXO ports, there are 2 ways to detect callee has already picked up the call, one is to detect reverse signal, and the other is to detect the termination of ring back tone, so user must set ring back tone of PBX or

PSTN network.
(If user doesn't know about the frequency of disconnect tone or ring back tone, please refer to ***[record]*** command to detect frequency.)
For each tone may have 1 set or 2 sets (high and low) of frequencies. If user wants to set 0 in on/off time, please set "1023" represent "0". (ex. ***tone 1 620 480 8 8 50 50 1023 1023***)
(***tone "index of tone" "frequency of high" "frequency of low" "level of high" "level of low" "on time of high" "off time of high" "on time of low" "off time of low"***)

5.20 [fxopwd]

This command is for FXO ports.

---

usr/config$ fxopwd

FXO password information and configuration
Usage:
fxopwd -add [passwd number][direction number]
fxopwd -delete index
fxopwd -modify index [passwd number][direction number]
fxopwd -print FXO password information.
passwd The password.
Example:
fxopwd -add passwd 1234
fxopwd -delete 1
fxopwd -modify 1 passwd 1234

usr/config$

Parameter Usages:

-print	show all FXO password configuration
-add	add 1 set of FXO password
-delete	delete 1 specific set of FXO password
-modify	modify 1 specific set of FXO password

passwd password

5.21 [record]

User can record greeting and askpin file and analyze tone frequency by calling in FXO line of Gateway.

---

usr/config$ record

Record greeting voice and ask pin code voice, tone analyze.
Usage:
record -greeting filename
-askpin filename
-tone
Example:
record -greeting greeting.100
record -askpin askpin.100
record -tone

usr/config$

Parameter Usages:

-greeting record greeting file. User must assign a file name for greeting, once record is finished, file recorded will be display in rom –print.

---

usr/config$ record -greeting test.100

Please off hook TEL 1 and press (N) for next step...
n

Press (R) to start record...
r

Press (S) to stop record...
.....
.....
.....s.....
.....
.....

Press (P) to play the voice or (W) to write to flash or (Q) to quit...
p
w

Please wait a moment...

Write flash ok...

Boot Rom : sdboot.200
Application Rom : fxso.100
DSP App : 48302ce3.300
DSP Kernel : 48302ck.300
DSP Test Code : 483cbit.bin
Greetings : test.100
Ask Pin : askpin.100
q
usr/config$

-askpin record askpin file. User must assign a file name for askpin file, once record is finished, file recorded will be display in rom –print.

---

usr/config$ record -askpin askpintest

Please off hook TEL 1 and press (N) for next step...
n

Press (R) to start record...
r

Press (S) to stop record...

.....
.....
.....
.....
.....
.....s.....
.....
.....
.

Press (P) to play the voice or (W) to write to flash or (Q) to quit...

p

w

Please wait a moment...

Write flash ok...

Boot Rom : sdboot.200
Application Rom : fxso.100
DSP App : 48302ce3.300
DSP Kernel : 48302ck.300
DSP Test Code : 483cbit.bin
Greetings : greeting.100
Ask Pin : askpintest

q

usr/config$

Note: Remember to press enter after press any command.

-tone analyze tone frequency. Gateway can analyze tone frequency as user provide tone in FXO Line1.

---

usr/config$ record -tone

Press (R) to start record...
r
.....
.....
.....
.....

Analyzing!! Please wait a moment……..
Frequency 1 : 480
Frequency 2 : 620
Frequency 3 (2623) is more than 1000, please ignore it.
usr/config$

Note:

1. ***Record ring back tone: user can use FXS Line1 to call FXO Line1, after hearing ring back tone, use this command to detect frequency of ring back tone.***
2. ***Record disconnect tone: Please read the procedure of recording disconnect tone file from the web site in application.***
3. ***The value of disconnect tone and ring back tone will not write in flash automatically. Please use the command in "tone" to write in the tone table.***

The Procedures of recording the disconnect tone

Before you start ：

A PSTN line which connect with the Line 1 port.
An analog phone connect with the Tel 1 port.
Configure Peer-to-Peer mode.

Please record the disconnect tone just follow the stage as below ：

1. Please enter the command before you record the disconnect tone：
   record –tone
2. Make a call from PSTN side into Line 1 port.
3. You will get a greeting when the call enter the gateway.

4. Pease dial the number of the Tel 1 port.
5. The phone will ring if the number you dial is correct.
6. Pick up the phone and make sure the call is connect.
7. Hang up the phone which is from PSTN side and Tel 1 port will get the disconnect tone.
8. When you get the disconnect tone from the phone set of the Tel 1 port, press <**R**> and <**ENTER**> buttons to start recording the disconnect tone.
9. Please hang up the phone if you get the message as below：*Analyzing!! Please wait a moment…*
10. There are three values you will get after analyzing. Please leave the value which is over 1000 Hz, this is not the frequency of disconnect tone.
11. Please put the frequency in the tone table just follow the command：*tone 4 420 680 8 8 25 25 50 50*

【Example-1】

(Make a call from PSTN to FXO port)
usr/config$ record -tone

Press (R) to start record...
(Please make sure that you are already finish the steps 2 ~ 7)
r　(Press “Enter” button after you key in “R”)
.....
.....
.....
.....

Analyzing!! Please wait a moment...
(You could hang up the call from PSTN if you get this message)

Frequency 1 : 481
Frequency 2 (2623) is more than 1000, please ignore it.
Frequency 3 : 621

tone 4 481 621 8 8 25 25 1023 1023
(Put this value in to the tone table)

tone –print

Disconnect tone 1 parameter
Frequency high : 620
frequency low : 480
frequency high level : 8
frequency low level : 8
Tone1 on : 25
Tone1 off : 25
Tone2 on : 1023
Tone2 off : 1023
Disconnect tone 2 parameter
Frequency high : 450
frequency low : 0
frequency high level : 8
frequency low level : 0
Tone1 on : 35
Tone1 off : 35
Tone2 on : 1023
Tone2 off : 1023
Disconnect tone 3 parameter
Frequency high : 620
frequency low : 480
frequency high level : 8
frequency low level : 8
Tone1 on : 50
Tone1 off : 50
Tone2 on : 1023
Tone2 off : 1023
Disconnect tone 4 parameter
Frequency high : 621
frequency low : 481
frequency high level : 8
frequency low level : 8
Tone1 on : 25
Tone1 off : 25
Tone2 on : 50

Tone2 off : 50
(Confirm the values is correct or not)

(Key in the commit and reboot command if you finish the procedures as above)
【Example-2】
(Make a call into FXO port)
usr/config$ record -tone

Press (R) to start record...
(Please make sure that you are already finish the steps 2 ~ 7)
r (Press “Enter” button after you key in “R”)
.....
.....
.....
.....

Analyzing!! Please wait a moment...
(You could hang up the call from PSTN if you get this message)

Frequency 1 : 473
Frequency 2 (2623) is more than 1000, please ignore it.
Frequency 3 (1856) is more than 1000, please ignore it.

tone 4 473 473 8 8 25 25 1023 1023
(Please configure the high and low frequency as the same value if you just get a signal frequency)

tone –print

Disconnect tone 1 parameter
Frequency high : 620
frequency low : 480
frequency high level : 8
frequency low level : 8
Tone1 on : 25
Tone1 off : 25
Tone2 on : 1023
Tone2 off : 1023

Disconnect tone 2 parameter

Frequency high : 450
frequency low : 0
frequency high level : 8
frequency low level : 0
Tone1 on : 35
Tone1 off : 35
Tone2 on : 1023
Tone2 off : 1023

Disconnect tone 3 parameter

Frequency high : 620
frequency low : 480
frequency high level : 8
frequency low level : 8
Tone1 on : 50
Tone1 off : 50
Tone2 on : 1023
Tone2 off : 1023

Disconnect tone 4 parameter

Frequency high : 621
frequency low : 481
frequency high level : 8
frequency low level : 8
Tone1 on : 25
Tone1 off : 25
Tone2 on : 50
Tone2 off : 50

(Confirm the values is correct or not)

(Key in the commit and reboot command if you finish the procedures as above)

5.22 [pt]

RTP payload type configuration and information

---

usr/config$ pt

RTP payload type configuration and information
Usage:
pt-print Display the RTP payload type information
-rfc2833 Configure the DTMF RFC2833 payload type
-dtmf Configure the DTMF payload type
-fax Configure the FAX payload type
-faxbypass Configure the FAX ByPass payload type
-modembypass Configure the MODEM ByPass payload type
-redundancy Configure the Redundancy payload type
-modemrelay Configure the MODEM Relay payload type

Example:
pt -rfc2833 96 -fax 101

usr/config$

5.23 [rom]

ROM file information and firmware upgrade function.

---

usr/config$ rom

ROM files updating commands
Usage:
rom [-print][-app][-boot][-dsptest][-dspcore][-dspapp][-greet][-askpin]
-s TFTP/FTP server ip -f filename
rom -print
-print show versions of rom files. (optional)
-app update main application code(optional)
-boot update main boot code(optional)
-boot2m update 2M code(optional)
-dsptest update DSP testing code(optional)
-dspcore update DSP kernel code(optional)
-dspapp update DSP application code(optional)
-greeting update greeting voice file(optional)

-askpin update ask pin code voice file(optional)
-s IP address of TFTP/FTP server (mandatory)
-f file name(mandatory)
-method download via TFTP/FTP (TFTP: mode=0, FTP: mode=1)
-ftp specify username and password for FTP
Note:
This command can run select one option in 'app', 'boot',
, 'dsptest', 'dspcore', and 'dspapp'.
Example:
rom -method 1
rom -ftp vwusr vwusr
rom -app -s 192.168.4.101 -f app.bin

usr/config$

Parameter Usages:

-print	show versions of all rom files
-app, boot, boot2m, dsptest, dspcore, dspapp, greeting, askpin	to update main Application program code, Boot code, DSP testing code, DSP kernel code, DSP application code, greeting file, askpin file.
-s	to specify TFTP server's IP address when upgrading ROM files.
-f	to specify the target file name, which will replace the old one.
-method	to decide using TFTP or FTP as file transfer server. "0" stands for TFTP, while "1" stands for FTP.
-ftp	if users choose FTP in above item, it is necessary to specify pre-defined username and password when upgrading files.

5.24 [passwd]

For security concern, users have to input the password before entering configuration mode. "**passwd**" command is for password setting purpose.

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usr/config$ passwd

Password setting information and configuration

Usage:

passwd -set Loginname Password

passwd -clean

Note:

1. Loginname can be only 'root' or 'administratoar'

2. passwd -clean will clear all passwd stored in flash, please use it with care.

Example:

passwd -set root Your_Passwd_Setting

usr/config$

Parameter Usages:

-set

(***passwd –set "login name" "password"***)

Note : "login name" can be "***root***" or "***administrator***" only. "root" and "administrator" have the same authorization, except some commands that can be executed by "root" only – "***passwd –clean***", "***rom –boot***", "***rom –bot2m***" and "***flash –clean***".